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On the relations among temporal integration for loudness, loudness discrimination, and the form of the loudness function. (A)

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MONDAY MORNING, 13 MAY 1996 CELEBRATION B, 8:00 TO 11:50 A.M.

Session 1aAO

Acoustical Oceanography: Inversions for Surface, Volume, and Bottom Properties

Nicholas C. Makris, Chair
Naval Research Laboratory, 4555 Overlook Avenue, N.W., Washington, DC 20375

Chair's Introduction—8:00

Contributed Papers

8:05

1aAO1. Estimating surface orientation from sonar images. Nicholas C. Makris (Naval Res. Lab., Washington, DC 20375)

Sonar images of remote surfaces are typically corrupted by signal-dependent noise known as speckle. This noise arises when wavelength scale roughness on the surface causes a random interference pattern in the sound field scattered from it by an active system. Relative motion between source, surface, and receiver cause the received field to fluctuate over time with complex Gaussian statistics. Underlying these fluctuations, however, is the expected radiant intensity from the surface, from which its orientation may be inferred. In many cases of practical importance, Lambert's law is appropriate for such inference because variations in the projected area of a surface patch, as a function of source and receiver orientation, often cause the predominant variations in its radiance. Therefore, maximum likelihood estimators for Lambertian surface orientation are derived. These are asymptotically optimal when a sufficiently large number of independent samples are available, even though the relationship between surface orientation and measured radiance is generally nonlinear. Here, the term optimal means that the estimate is unbiased and its mean square error equals the Cramer–Rao lower bound, which is also derived. The requisite number of independent samples necessary for asymptotic optimality of the maximum likelihood estimate is given for some special cases.

8:20

1aAO2. Broadband frequency response modeling for inversion of a shallow-water waveguide with a range-dependent soft clay bottom (Yellow Shark '95 experiment). Raymond J. Soukup (Naval Res. Lab., Washington, DC 20375-5320), Jean-Pierre Hermand, and Enzo Michelozzi (SACLANT Undersea Res. Ctr., 19138 La Spezia, Italy)

During the Yellow Shark '95 experiment the broadband frequency response (200–800 Hz) of a shallow-water waveguide was measured for the purposes of performing multimodal inversion of its acoustic and geoacoustic properties. Measurements from vertical receiving arrays deployed every 8 km along a 40-km tomography transect showed that the frequency response was mostly flat with high transmission loss at selected frequencies. This differs from often observed frequency dependence in other shallow-water environments where there is some low frequency with minimum loss, suggesting an optimum frequency of propagation. Range-dependent geoacoustic and seismic data along the transect and water sound-speed data measured during the acoustic transmissions permitted the development of a realistic environmental model. Normal-mode modeling results with C-SNAP indicated that the presence of a soft clay top layer with a speed less than the water column was a decisive factor in the observed frequency dependence, since it acted as an alternative waveguide to lower-order modes. The modeled frequency dependence and absolute level of transmission loss closely agreed with the measurements obtained at the different ranges and depths, resulting in a validation of theoretical arguments involving selected frequencies of high transmission loss as a major feature of this type of environment.

8:35

1aAO3. Inversion techniques for characterizing a coupled mode acoustic waveguide. Daniel Rouseff and Martin Siderius (Appl. Phys. Lab., Univ. of Washington, Box 355640, Seattle, WA 98105)

A method for characterizing a range-dependent acoustic waveguide by remote sensing is examined. The approach is to transmit from a limited number of source locations and measure the propagating field along a vertical array. Characterization of the environment is cast as an inverse problem where the goal is to determine the mode coupling coefficients from the acoustical measurements. Ideally, measurements from the source at a single location should suffice to characterize the medium when the propagation is adiabatic. When there is coupling, the problem is underdetermined and unstable if the number of source locations is less than the number of propagating modes. Tikhonov regularization is used to build stability into the inversion algorithm. The number of measurements required is shown to be directly related to the nature and extent of the mode coupling. The method is demonstrated by numerical simulation for two shallow water scenarios: a sloping bottom and a randomly rough bottom. [Work supported by ONR.]

8:50

1aAO4. Time-domain shallow-water environmental inversion using unknown broadband sources. Jacob Roginsky (Naval Res. Lab., Washington, DC 20375)

Inversion for environmental parameters using both simulated and actual experimental data was performed based on the propagation results for presumably unknown broadband sources. The environment is a shallow-water environment and all computations are performed in the time domain. The inversion is based on the least-square comparison between the results of a source deconvolution for each of several receivers. Because single-receiver source deconvolution in underwater acoustics is an intrinsically ill-posed problem, the Tichonov–Philips regularization is incorporated into the current procedure. The method is made efficient by the use of a new deconvolution algorithm that specifically takes advantage of the triangular Toeplitz structure of the propagation matrix [J. Roginsky and G. W. Stewart, J. Acoust. Soc. Am. **97**, 3290 (A) (1995)]. The advantages of this method are: (a) The process can be performed with as few as two receivers; (b) no *a priori* source information is required; (c) due to one's ability to work with shorter time windows, the procedure can be made faster than a corresponding frequency domain approach. [Work supported by ONR.]

9:05

1aAO5. Robust ray path identification for ocean acoustic tomography. Max Deffenbaugh (Dept. of Elec. Eng. and Comput. Sci., MIT, Cambridge, MA 02139), Henrik Schmidt (MIT, Cambridge, MA 02139), and James G. Bellingham (MIT Sea Grant, Cambridge, MA 02139)

Acoustic tomography in a ray environment is a two-step process. First, each measured ray arrival time in the received multipath structure is identified with a particular predicted ray path. Second, the differences between the measured arrival times and the predicted arrival times for the represented ray paths are used in a linear inversion to calculate corrections to parameter values describing the sound-speed structure. If measured arrival times are identified with the wrong ray paths, errors will result in the linear inversion. In deep ocean tomography, the time spacing between ray arrivals is typically large compared to the parameter induced changes in arrival times, so ray path identification is not difficult. In shallow water, however, ray path identification can be more challenging. An algorithm is presented that combines the ray path identification step with the linear inversion step to allow tomography using rays where the parameter induced arrival time shifts may be larger than the time spacing between arrivals. The performance of the algorithm is simulated for a typical shallow-water environment. [Work supported by ONR.]

9:20

1aAO6. Second-order analysis of eigenray displacement: When does Fermat's principle apply? B. Edward McDonald, Michael D. Collins (Naval Res. Lab., Washington, DC 20375), and Ira B. Bernstein (Yale Univ., New Haven, CT 06520)

In the ocean sound channel multiple eigenrays, each possessing stationary travel time, may connect two fixed points. In a highly structured propagation environment, how does one determine if the travel time of a given eigenray is a local minimum (Fermat's principle), maximum, or saddle point with respect to ray parameters? A second-order analysis of the travel time integral along an eigenray is carried out to reveal conditions determining the nature of the stationarity. In three dimensions, the problem reduces to a second-order differential/diadic eigenvalue equation along the ray. In two dimensions, it becomes a scalar Sturm–Liouville eigenvalue problem. An analytic example is given in which both maximum and minimum travel time eigenrays exist. Conditions for ray chaos in the analytic example are derived and related to the occurrence of maximum travel time eigenrays. [Work supported by NRL.]

9:35

1aAO7. Computer modeling of the process of the ocean stream velocity remote acoustics sensing in Fram Strait environmental condition. Igor B. Esipov (N. Andreyev Acoust. Inst., 117036, Moscow, Russia), Ola M. Johannessen (Nancen Environ. and Remote Sensing Ctr., N-5037 Solheimsviken-Bergen, Norway), Konstantin A. Naugolnykh (CIRES/Environ. Technol. Lab., Boulder, CO), Oleg B. Ovchinnikov, Yury I. Tuzilkin, and Viktor V. Zosimov (N. Andreyev Acoust. Inst., 117036, Moscow, Russia)

An important part of the general problem of global climate change is the measurement of the heat inflow from the Atlantic ocean to the Arctic basin. An effective tool to measure it is the acoustic methods application. Some aspects of this problem connected with the average water temperature measurements were considered previously [Naugolnykh *et al.*, J. Acoust. Soc. Am. **97**, 3264(A) (1995)]. In the present paper the results of computer modeling of the ocean stream velocity remote acoustical sensing with respect to the Fram Strait environment are presented. The computer simulation of the random environment characterized by the 3-D sound-speed field was performed. This process has been done with the assumption of both regular temperature changing in the cross section of the Fram Strait and isotropic random sound-speed variation distribution in the plane in each horizon. Spatial spectra of simulated inhomogeneities can be corrected in the course of modeling. Simulated fields of environmental inhomogeneities were used for the sound signal propagation modeling through the paths directed crosswise and along the water flow to compare Doppler and scintillation methods of stream monitoring.

9:50–10:05 Break

10:05

1aAO8. Mode coupling effect of a cold eddy in Taiwan's Northeast Sea. Chi-Fang Chen (Dept. of Naval Architecture and Ocean Eng., Natl. Taiwan Univ., Taiwan, R.O.C.)

The Kuroshio current turns toward the north-east while it passes the Okinawa Trough in Taiwan's Northeast Sea. In this region, there are different waters mixing, namely, water from the Taiwan Strait, water of the East China Sea, and the Kuroshio current. During the year, except for the summer months, the Kuroshio current is characterized by its distinct temperature difference from other waters, i.e., the Kuroshio front and cold eddies are two predominant features in this region. The frontal effect on acoustic propagation has been studied and reported in a previous talk. In this paper, the coupling effect of modes when acoustic signals transmit through the cold eddy using both the Parabolic Equation method and the Adiabatic Mode Approximation is reported. Severe mode coupling is realized while the sound transmits through the cold eddy. [Work supported by National Science Council of Republic of China.]

10:20

1aAO9. Multimodal dispersion identification. James Poplawski and Ted Birdsall (Dept. of Elec. Eng. and Comput. Sci., Univ. of Michigan, 1301 Beal, Ann Arbor, MI 48109)

Geometric dispersion in the deep ocean sound channel causes wide-band modal arrivals to be spread significantly in time at long ranges. This results in a low modal peak signal-to-noise ratio and interference due to multiple overlapping arrivals. These effects make the identification of individual modal arrivals difficult without the use of a vertical receiving

array. The time compression of modal arrivals using phase-only filters is examined as a method for locating and identifying arrivals from a reception at a single depth. A search display, dubbed the dispersion diagnostic display, has proved its worth in computer simulations using a 60 mode range invariant model of SLICE89 propagation. The DD display simultaneously shows a raylike (no dispersion) structure and a patterned modal structure with strongly dispersed early arrivals and lightly dispersed terminal arrivals. [Work supported by ATOC.]

10:35

1aAO10. Improved empirical orthogonal functions for ocean acoustic tomography. Max Deffenbaugh (Dept. of Elec. Eng. and Comput. Sci., MIT, Cambridge, MA 02139) and Henrik Schmidt (MIT, Cambridge, MA 02139)

The method of empirical orthogonal functions is commonly used in tomography to select a set of basis vectors to represent variations in the sound-speed profile. In this method, historical profiles are used to estimate a profile covariance matrix, and the eigenvectors corresponding to the largest eigenvalues of the covariance matrix are taken for the basis vectors. These basis vectors are the most efficient parametrization of the variations in the profile. They are NOT, however, the parametrization which leads to the most accurate post-measurement estimate of the profile, because, in a tomographic experiment, all profile variations cannot be measured with the same accuracy. In this paper, the set of basis vectors, which will yield the least error in the post-measurement estimate of the profile, are derived taking into account measurement resolution and measurement noise. These improved empirical orthogonal functions are applied to several canonical problems in ocean acoustic tomography, and the resulting enhancement in estimation accuracy is demonstrated. [Work supported by ONR.]

10:50

1aAO11. Estimation of stochastic wave-number variations at the Atlantic Generating Station site and comparison with acoustic data. Michael J. Longfritz (Rensselaer Polytechnic Inst., Troy, NY 12180-3590), Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), William L. Siegmann, Melvin J. Jacobson, and Indra Jaya (Rensselaer Polytechnic Inst., Troy, NY 12180-3590)

The estimation of horizontal wave-number variations in a stochastic shallow-water environment at the Atlantic Generating Station (AGS) site is constructed. A method using empirical orthogonal functions has been developed [Longfritz *et al.*, J. Acoust. Soc. Am. **97**, 3316 (A) (1995)] to relate wave-number fluctuations to environmental variations. First, the AGS site is modeled as a stochastic shallow channel with a range-dependent layered sediment bottom. The layer interface depths and intralayer sound speeds are treated as random variables. Then, the estimation procedure is applied to a particular propagation track, where geoacoustic profiles and acoustics measurements are both available. Environmental profiles along the track are treated as a sample from a stochastic ensemble. Estimates are obtained for both the range variance of the wave numbers and the particular deviations associated with each profile in the sample. Comparisons are made between results from the estimation procedure and simulations from the KRAKEN propagation code. In a separate study, acoustical wave numbers have been generated from experimental data for the track using a method of Thomson. The estimation results are compared to this data.

11:05

1aAO12. Acoustic inversion of bottom reflectivity and bottom sound-speed profile in shallow water. T. C. Yang (Naval Res. Lab., Washington, DC 20375) and T. Yates (Vector Res. Co., Inc., Rockville, MD 20852)

This paper presents a full field technique to invert bottom sound profile and bottom reflectivity from acoustic data collected in shallow water. Bottom sound-speed profile and bottom reflectivity have been traditionally estimated using seismic reflection/refraction when acoustic ray paths and travel time can be identified and measured from the data. Due to the shallow depth in shallow water, the many multipaths due to bottom reflection/refraction make such identification and measurement rather difficult. A full field inversion technique is presented which uses a broadband source and an easily deployable array for bottom sound speed and reflectivity inversion. The technique is a modified matched-field inversion technique referred to as matched beam processing. Matched beam processing uses conventional beamforming processing to transform the field data into the beam domain and correlate that with the replica field also in the beam domain. This allows the analysis to track the acoustic field as a function of incidence/reflected angle and minimize contamination or mismatch due to sidelobe leakage. This technique is illustrated with simulated data for a vertical array and can be generalized to other array configurations, such as a towed array.

11:20

1aAO13. Determination of sub-bottom sediment properties and their spatial distributions from chirp sonar data. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

Determination of impedance and attenuation coefficients for upper sediment layers from chirp sonar data is performed by using simulated annealing algorithms. Normal incidence reflection seismograms are calculated for layered marine sediments using the Biot theory of acoustic wave propagation in porous media to produce synthetic data. A time-domain inversion algorithm progressively identifies each reflection and applies random perturbations to corresponding model parameters to minimize the quadratic deviation between the data and synthetic seismograms. Numerical simulations indicate that such an inversion scheme can be efficiently used to survey large areas for characterization of sediment properties and their spatial distributions. In addition, impedance and attenuation estimates are used to predict the sediment porosity, density, permeability, and sound speed using the Biot theory. Finally, chirp sonar data (3 to 7 kHz) collected on the New Jersey Shelf as a part of SWARM-95 (Shallow Water Acoustic Random Media) experiment are analyzed. Inversion results confirm previous coring and *in situ* observations that alternating sand and clay layers are present at the Hudson Apron site.

11:35

1aAO14. Radiation impedance of a circular vibrating plate on the surface of viscoelastic media. Masao Kimura (Dept. of Ocean Eng., Tokai Univ., 3-20-1 Orido, Shimizu, Shizuoka, 424 Japan)

Radiation impedance of a vibrating plate on the surface of a viscoelastic medium such as marine sediment varies with the viscoelastic properties of the medium. Therefore, it is possible to predict the viscoelastic properties using the characteristics of the radiation impedance. In this paper, the characteristics of radiation impedance of a circular vibrating plate on the surface of viscoelastic media were obtained by numerical analysis. The results showed that the values of ka (k is the wave number, a is the radius of a circular vibrating plate) at which the imaginary part of the radiation impedance becomes zero can be related to the ratio of the shear wave velocity to the longitudinal wave velocity, that is, the Poisson's ratio. Next, measurements of radiation impedance for three kinds of urethane rubbers and two kinds of dry beach sands were done, and the measured values were compared to the calculated values.

Session 1aSC

Speech Communication: Informal Workshop on Voice Perception I

David B. Pisoni, Cochair
Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, Indiana 47405

Jody Kreiman, Cochair
University of California, Los Angeles, Head/Neck Surgery, 31-24 Rehab, Los Angeles, California 90024-1794

The goal of this informal workshop is to bring together several researchers working on various aspects of voice perception. Historically, the study of voice has been treated as a more-or-less autonomous area quite distinct from other research problems in speech and hearing sciences. In this workshop, some of the traditional problems of voice classification and perception will be discussed and reviewed and then these efforts will be related to recent findings in speech perception and spoken word recognition which have shown important dependencies between traditional voice parameters and perceptual analysis of the speech signal. The contributors to the session include: Jody Kreiman, Diana VanLancker, Keith Johnson, Robert Remez, Peter Jusczyk, and David Pisoni. Please bring your slides and overheads and join in the discussion on some long-standing problems related to the "indexical" and "linguistic" functions of speech. For further information about this informal workshop, please feel free to contact D. B. Pisoni via E-mail: pisoni@indiana.edu. Everyone with an interest in speech perception, production, and spoken language processing is welcome to attend and contribute to the discussion of these problems.

MONDAY AFTERNOON, 13 MAY 1996 CELEBRATION B, 1:00 TO 5:35 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Ambient Noise and Inversions in the Surf Zone

Grant B. Deane, Chair
Marine Physical Laboratory, Scripps Institution of Oceanography, La Jolla, California 92093-0238

Chair's Introduction—1:00

Invited Papers

1:05

1pAO1. Ambient noise from the surf zone: Surface and acoustic waves. W. Kendall Melville (Scripps Inst. Oceanogr., U.C.S.D., La Jolla, CA 92093-0213)

Wave breaking is probably the dominant source of ambient noise in the ocean. In deep water, breaking is intermittent in both space and time, but, in the surf zone, every wave of any appreciable size breaks, or shoals, before it reaches the shore. This makes the surf zone and its neighboring region offshore acoustically different from deeper water. In shoaling, surface waves dissipate their energy, generate currents and turbulence that mix the water column, suspend sediments and entrain bubbles, and generate sound. It is the potential relationships between the sound generation and propagation and the other physical processes associated with surface waves, and the shoaling bottom topography, that is the subject of this talk. This is an open field in acoustical oceanography and underwater acoustics, with few measurements available but a great deal of potential. In this lecture the processes associated with the shoaling of surface waves will be reviewed and the resulting implications for sound generation and propagation will be discussed. The potential for the inversion of acoustic data to infer other surf zone processes will be explored.

1:25

1pAO2. Recent results in surf zone ambient noise measurements. Ellen S. Livingston (Office of Naval Res., Code 321, 800 N. Quincy St., Arlington, VA 22217-5000)

Over the years only a few measurements have been made of surf zone ambient noise. Recently, however, several sets of data have been taken in and near the surf zone in conjunction with coastal oceanography and seismoacoustic studies. In this talk, an overview is given of past and present surf zone studies and differing approaches are discussed and contrasted. The variety of results now starting to come forward from the new data are outlined. Important issues opened by these recent results and new applications anticipated are discussed as well.

1pAO3. Surf noise and biological sounds in the near-surf-zone environment. Gerald L. D'Spain, William S. Hodgkiss, William A. Kuperman, and Lewis P. Berger (Marine Physical Lab, Scripps Inst. of Oceanogr., San Diego, CA 92106)

The Marine Physical Lab's Adaptive Beach Monitoring Experiment was conducted from 24 April to 14 June 1995, in and near the surf zone off the S. California coast. Other participants included NRaD/NCCOSC, NRL, ARL/PSU, and ARL/UT. The focus here is on the underwater acoustic noise field properties. Contributions of breaking surf are studied in the 1- to 400-Hz band using a 35-element subsection of a bottom hydrophone array oriented perpendicular to the coast and located 3.4-km offshore in 20-m water. A 3-week time series of endfire-beamforming results are compared to the ocean surface wave height data collected over the same period. Source tow data quantify the propagation effects of surf noise to the array site and permit inversions for bottom geoacoustic properties. The array data are supplemented by the distance dependence from shore of data from bottom-mounted sonobuoys in the 100-Hz–20-kHz band, and air acoustic data recorded just landward of the shoreline. Biologics are a substantial component of the noise field and must be accounted for in evaluating the contribution from surf. Fish sounds themselves can be used as novel sources in inversion for bottom geoacoustic properties. [Work supported by ONR, Code 32.]

2:05

1pAO4. Matched-field inversion in a rapidly fluctuating shallow-water waveguide. Nicholas C. Makris (Naval Res. Lab., Washington, DC 20375)

A number of useful applications for matched-field inversion have been envisioned for the littoral zone. These include the estimation of environmental parameters, such as the sound-speed structure of the water column and geoacoustic properties of the sediment, as well as the more traditional localization of submerged objects. However, natural disturbances such as passing surface and internal gravity waves often place shallow-water waveguides in such a state of flux that a signal, deterministic when transmitted from a source, becomes fully randomized after propagating only several channel depths away in range to a receiver. When not properly accounted for, such randomization can severely degrade the accuracy of a matched-field inversion, which presumably is for parameters that remained fixed during the measurement process. Therefore, the general performance of a matched-field inversion is examined from the perspective of statistical estimation, under the worst case scenario of a waveguide that may be fluctuating rapidly during a given transmission or set of transmissions. Specific examples for shallow-water matched-field tomography and object localization using ambient noise are then considered.

2:25

1pAO5. Shallow-water ambient noise caused by breaking waves in the surf zone. Oscar B. Wilson, Marc S. Stewart[a] (Naval Postgrad. School, Dept. of Phys., Monterey, CA 93943), James H. Wilson (Neptune Sciences, Inc., San Clemente, CA 92674), and Robert H. Bourke (Naval Postgrad. School, Monterey, CA 93943)

Ambient noise measurements made by Wilson, Wolf, and Ingenito in Monterey Bay, California in 1980 and 1981 were combined with modeled transmission loss (TL) to estimate the spectral source level of surf-generated noise. In the frequency range 20–700 Hz the horizontal directionality of these noise data showed much higher levels from shore than seaward and the overall noise levels decreased with range from shore, strongly suggesting that surf noise was a significant contributor. A Hamilton geoacoustic model of the coastal environment has been derived and used in a finite element parabolic equation propagation loss model to obtain the TL values. These indicate that propagation is significantly dependent on the environment and frequency, particularly near the surf zone. Estimates of wind and local wave noise intensity corrected for the local environment were subtracted from the total measured noise field to determine the contribution due to surf only. Heavy surf breaking on a uniform 12.5-km linear section of beach near Fort Ord was found to be the dominant source of surf-generated noise. The preliminary results provide estimates of the noise source level densities for heavy and light surf at Fort Ord beach. [a]Currently at Naval Sea Systems Command, Arlington, VA 22242-5160.

2:45–3:00 Break

Contributed Papers

3:00

1pAO6. Imaging of passive targets in shallow-water using an ambient noise acoustic imaging system. Marc P. Olivieri, Stewart A. L. Glegg, and Robert K. Coulson (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

This paper will describe an experiment on the detection of passive targets in shallow water using ambient noise as the sole source of insonification. Glegg and Olivieri [Noise-Con 95, p. 1179] proposed a broadband signal processing technique to achieve this type of passive acoustic detection using a limited number of transducers. An array has been developed to apply this technique in a shallow-water environment. This paper will describe an experiment carried out in Summer 1995 off the Boca Raton inlet in which a target was successfully detected. The size of the target chosen for this test was 1.8×1.2 m and it was placed 40 m from the array. The acoustic field was imaged with and without the target present and both passive and active detection of the target was obtained. The measurements were carried out in low sea states when the primary source of ambient noise was biological and strongly anisotropic. [Work supported by ONR.]

3:15

1pAO7. Inversion of the underwater sound field to quantify the formation of a subsurface bubble layer. Jeffrey A. Nystuen (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The underwater sound field can be used to quantify oceanic processes that actively produce sound, including, for example, wind/wave breaking (wind speed) and rainfall. These "active" sound sources are subsequently modified by the acoustic environment. In the case of wave breaking or rainfall, the sound source is located at the ocean surface. One of the principal features of the near-surface zone in the ocean capable of modifying the sound field is the ambient bubble field. Frequency-dependent modifications of the underwater sound field during heavy rainfall are observed. Using a rainfall generated sound field as an example, the change in the spectral character of the sound field will be used to quantify the formation and decay of an ambient bubble field. Documentation of the formation of a bubble layer has implications for gas and momentum exchange at the ocean surface by rainfall.

3:30

1pAO8. Acoustic measurements of breaking waves in the surf zone. Grant B. Deane (Scripps Inst. of Oceanogr., Mail Code 0238, La Jolla, CA 92093-0238)

A series of measurements have been made to characterize the sound generated by breaking waves in the surf zone around Scripps Pier. Simultaneous video and acoustic data were made to allow the acoustic energy radiated by individual waves to be correlated with the breaking event at the surface. The acoustic measurements were made directly below breaking waves in 2 m of water and 170 m from the shoreline. Spectral analysis of the data demonstrates the presence of a strong acoustic field below the breaking waves. Overhead waves generate sound for some several seconds, and are clearly distinguishable from nearby surf. The breaking wave spectrum is bimodal, with two peaks in intensity around 30 and 1000 Hz. A preliminary analysis of the data shows an acoustically active region of water propagating with the breaking wave. [Work supported by ONR.]

3:45

1pAO9. Preliminary characterization and modeling of ambient noise due to high surf during DUCK94. J. Paquin-Fabre (Neptune Sciences, Inc., 150 Cleveland Ave., Slidell, LA 70458) and James H. Wilson (Neptune Sciences, Inc., San Diego, CA 92123)

Broadband noise due to surf that propagates kilometers out to sea with little transmission loss has been observed for many years, but rarely directly measured and characterized. In October of 1994 a coastal research experiment, DUCK94, was conducted off the coast of Duck, North Carolina. Neptune Sciences, Inc., under Naval Research Laboratory sponsorship, participated in that experiment with the objectives of obtaining ambient noise data at offshore positions for investigations of nearshore acoustic noise generated by waves and surf. Measurements were taken, using modified Navy standard sonobuoys, primarily during high surf events (wave heights of 1.7 to 2.4 Hmo,m). Preliminary analysis of the data indicates a broadband noise maxima between 400 and 600 Hz. In order to characterize ambient noise due to surf for modeling purposes, source level densities for heavy surf are currently being developed using the measured AN data and a finite element parabolic equation transmission loss model.

4:00

1pAO10. Ambient noise measurements in the acoustic beach monitoring experiment. W. K. Melville, G. B. Deane, R. M. Shear, E. Terrill, and C. L. Epifanio (Scripps Inst. of Oceanogr., U.C.S.D., La Jolla, CA 92093-0213)

During the spring of 1994 (April–June) the Marine Physical Laboratory/SIO conducted the acoustic beach monitoring (ABM) pilot experiment at Camp Pendelton, CA. One component of the experiment was concerned with measurement of ambient noise up to 40 kHz, at distances up to several kilometers offshore. An array of three modified SSQ 57B sonobuoys were moored at distances from 1.5 to 3.5 km offshore. The surf zone was instrumented with pressure gauges and a current meter, and a comprehensive meteorological station was set up on the beach. A directional microphone on the beach was used to monitor the surf noise. Supporting oceanographic and meteorological data was available from the SIO coastal data information program. Near-shore acoustic spectra are characterized by two broad peaks in the ranges of 100–500 Hz and 1–7 kHz, respectively. The higher frequency peak rapidly attenuates offshore, while the lower frequency peak is essentially unattenuated over ranges of several kilometers. In contrast to deep water, the correlation of surf zone ambient noise with the local wind is poor since the shoaling waves may be dominated by swell from distant storms. Short term averages of acoustic power correlate very well with the amplitude modulation (groupiness) of the incident waves. [Work supported by ONR.]

4:15

1pAO11. Global directivity measurements of near shore ambient noise. Stewart A. L. Glegg, Marc P. Olivieri, and Robert K. Coulson (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

In the summer of 1995 a shallow-water ambient noise experiment was carried out 1-mile offshore of the Boca Raton Inlet. A volume array was deployed that provided both vertical and horizontal directivity measurements of the ambient noise. This paper will describe the array processing algorithm used to obtain the global directivity measurements with high resolution and present the ambient noise measurements. The results showed that the ambient noise was strongly anisotropic and dominated by biological noise from a local fishing pier located approximately 1 mile to the south. Measurements were also made of local boat traffic and it was found that the vertical arrival structure could be separated into clearly defined beams associated with each bottom bounce along the propagation path. [Work supported by ONR.]

4:30

1pAO12. Ambient noise generated by surface wind waves—Numerical and experimental results. Jin-Yuan Liu and Chuen-Song Chen (Dept. of Marine Environment, Natl. Sun Yat-sen Univ., Kaohsiung, Taiwan 804, R.O.C.)

Using a previously developed noise-generation model for sources due to surface processes [W. A. Kuperman and F. Ingenito, J. Acoust. Soc. Am. **67**, 1988–1996 (1980)], a numerical study of ambient noise, which incorporates realistic wind wave spectrum (e.g., Pierson–Moskowitz), was made for seismoacoustic ocean wave guides. The results demonstrate a close relationship between the noise field and the source field, allowing one to gain insight to the interplay of both fields. Experimental data of ambient noise and wind waves were collected in an experimental site near the central part of Taiwan. The measured data were also used to generate numerical results of ambient noise which are then compared with experimental data. The comparison has shown a reasonable agreement in the frequency range dominated by the surface wind waves for a receiver not too close to bottom. By modeling the marine environment as a shallow-water seismoacoustic wave guide, it was demonstrated that the noise model of the noise field was greatly affected by the sediment properties. [Work supported by National Science Council of Republic of Chian (Taiwan).]

4:45–4:50 Break

4:50–5:35 PANEL DISCUSSION

Panel Moderator: Grant B. Deane
Panelists: Gerald L. D'Spain, Ellen S. Livingston, Nicholas C. Makris, W. Kendall Melville, Thomas G. Muir, Oscar B. Wilson

Session 1pMU

Musical Acoustics: Nonlinear Effects in Wind Instruments

Douglas H. Keefe, Chair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, Nebraska 68131

Chair's Introduction—1:10

Invited Papers

1:15

1pMU1. Time-domain and frequency-domain explorations of nonlinearity in musical wind instruments. Neville H. Fletcher (Australian Natl. Univ., Canberra, Australia)

Nonlinearity is an essential feature of the operation of sustained-tone musical instruments, as well as being an important feature in many impulsively excited instruments. Analysis of the behavior of wind instruments can be approached either through the time domain, in which case the linear resonant parts of the instrument are characterized by their impulse response, or in the frequency domain, in which case the characterization is in terms of mode shapes and frequencies. The generator mechanism in some cases has an embedded linear mode structure as well, but there is an essentially nonlinear coupling between it and the main instrument resonator. This paper examines the different types of information, or the complementary views of the same information, that can be obtained by time-domain and frequency-domain approaches to wind instrument acoustics.

1:45

1pMU2. Inference of nonlinear effects from spectral measurements of wind instrument sounds. James W. Beauchamp (School of Music and Dept. of Elec. Eng., Univ. of Illinois, Urbana, IL 61801)

That wind musical instruments in conjunction with their players are highly nonlinear can be inferred from measurements of their output spectra. These show that, for a given performed pitch, bandwidth generally increases with increasing rms amplitude. The cause of this harmonic generation is generally attributed to a nonlinear pressure/flow excitation mechanism which interacts with the instrument's resonant input impedance characteristic. Since there are no analytical solutions which predict mouthpiece spectra, one must rely on simulations and empirical measurements. Meanwhile, the pipe portion of the wind instrument has long been assumed to be a linear resonant system acting as a high pass filter with resonant minima corresponding to the input impedance maxima. This at least is true at low amplitudes. However, for the trombone large discrepancies in the filter response have been found between measurements made with sine wave input and those made under actual performance conditions, particularly above resonant cutoff. The general effect is that spectral bandwidth (and attendant tonal brightness) increases even more with increasing amplitude than if the system were linear. There is now evidence that this may be due to shock wave formation in the pipe of the instrument.

2:15

1pMU3. Wind instruments: Some unanswered questions. Cornelis J. Nederveen (Acacialaan 20, 2641 AC Pijnacker, The Netherlands)

The basics of functioning and tuning of wind instruments is reasonably explained by small-amplitude descriptions. Some details must be cleared up, such as corrections at the hole/bore transition, bends, and radiation at hole ends. In some cases, however, the linear description fails. For example, a clarinet becomes unplayable when its usual holes are replaced by much smaller and shorter ones (of the same impedance) [D. H. Keefe, J. Acoust. Soc. Am. **73**, 1804–1820 (1983)]. Since it was noted that this tube could be blown with another (i.e., a saxophone) mouthpiece, it was supposed that the superimposed static flow influences the tube impedance. This assumption was provisionally confirmed by measurements. Understanding this phenomenon may help to improve the ease of playing of the lowest notes on saxophones. The common quantitative description of the flute action does not explain why certain fork fingerings cannot be blown and why the "reed-flute" has a dominating fifth harmonic in its initial transient. To contribute to the discussion on the influence of wall vibrations on tone quality, a mechanism will be proposed which is possibly responsible for a coupling between air and wall vibrations.

2:45

1pMU4. The measured vocal tract impedance for clarinet performance and its role in sound production. Teresa D. Wilson (School of Music, Box 353450, Univ. of Washington, Seattle, WA 98195)

The role of the vocal tract in clarinet performance has been investigated by examining the vocal tract impedance, Z_u. Theory predicts that Z_u peaks aligned with harmonics of the pitch frequency stabilize the oscillation. The vocal tract impedance was measured directly with a one-microphone technique and indirectly from a calculation using the continuity of flow equation $p_u/Z_u+p_d/Z_d=0$, where p_u and p_d are the mouth and mouthpiece pressures and Z_d is the instrument impedance. The continuity equation was verified for single tones and then used to examine the role of the vocal tract in a variety of musical situations. In orchestral excerpts, the performer tends to align resonances with the first and/or second harmonics, and there is some dependence on the musical context. For multiphonics, the performer creates a resonance that supports an oscillation at a linear combination of the audible pitch frequencies. For tones played with pitchbend, induced by slacking the jaw muscles, the playing frequency is controlled by the frequency of a large-amplitude vocal tract resonance, for which $|Z_u| \approx 100-400$ CGS ohms. Z_u changes only slightly when clarion tones are played without the register key, suggesting that other variables are involved.

3:15

1pMU5. Understanding the operation of auxiliary fingerings on conical double-reed instruments. Robert H. Cronin (360 Marmona Dr., Menlo Park, CA 94025) and Douglas H. Keefe (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Players of the modern bassoon are accustomed to using ornate fingerings to stabilize and correct certain notes that are difficult to tune. These "auxiliary" fingerings open or close tone holes far below the first open tone hole. The primary subject of the investigation was a modern German-system bassoon. Calculations of impedance versus frequency were made using a program based on the Plitnik–Strong model of 1979, with several extensions. The model, in turn, was validated by comparison with measurements taken with a system developed by Keefe *et al.* The authors also investigated the behavior of auxiliary fingerings on a replica baroque bassoon, and sorted out a puzzling "feature" of a replica Renaissance alto shawm. Based on this work, criteria for "good" note behavior are postulated. [Work supported by the International Double Reed Society.]

3:45–4:00 **Break**

Contributed Papers

4:00

1pMU6. A linearized model of bassoon sound production: The role of auxiliary fingerings. Douglas H. Keefe (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131) and Robert H. Cronin (Menlo Park, CA 94025)

A quantitative theory of sound regeneration in the bassoon helps explain differences in the playing frequency and mouthpiece spectra associated with the use of auxiliary fingerings. The linearized condition for steady-state oscillations at a frequency f is $0=Y(f)+Y_G(f)$; the air-column input admittance is $Y(f)$ and the double-reed generator admittance is $Y_G(f)$, whose real part is negative (Fletcher, 1979; Thompson, 1979). The generator admittance is based upon a simple Bernoulli-type flow model, and the cane reed is modeled as a damped oscillator. Model parameter values are empirically determined and combined with measured input admittances, allowing direct comparison of $Y(f)$ and $-Y_G(f)$ at frequencies below and above the open tone-hole lattice cutoff frequency (Benade, 1960). Auxiliary fingerings produce significant differences in input admittance magnitude and phase above cutoff, and sound production is stabilized for a fingering such that the linearized condition is satisfied near a harmonic multiple of the playing frequency. The model accounts for intonation shifts associated with changes in auxiliary fingerings. While sound production in the bassoon is highly nonlinear, the fine structure of the linear air-column response at frequencies above cutoff is an essential contributor. [Work partially supported by the International Double Reed Society.]

4:15

1pMU7. Time-domain simulation of an organ flue pipe. Seiji Adachi (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seika-cho, Kyoto 619-02, Japan)

Due to the difficulty in treating jet behavior, a general agreement on the sound production of flue instruments has not yet been obtained. This paper discusses three jet deflection models proposed by several authors by means of the time-domain approach. Simulations of an organ flue pipe where the blowing pressure is changed as a parameter are carried out. These models in common provide oscillations in the fundamental pipe resonance mode and in the higher modes (overblows). They also provide the underblow regimes of oscillation when the blowing pressure is lower. Differences among the models appear in the number of the underblow regimes and in the levels of harmonics of oscillations. Transition among the oscillation modes would also be a clue to compare the models. The results should be compared with an experiment of the artificial blowing of a pipe in the wide range of blowing pressure, which is in progress.

4:30

1pMU8. Influences of sharp toroidal bends on the tuning of wind instruments. Cornelis J. Nederveen (Acacialaan 20, 2641 AC Pijnacker, The Netherlands)

A toroidal bend causes tuning changes, which are mode dependent and may amount to 10 cents. This is caused by a change of the inertance in the bend, whereas the compliance is not affected. The inertance change was studied by Nederveen [*Acoustical Aspects of Woodwind Instruments* (1969)], Rostafinski [J. Acoust. Soc. Am. **52**, 1411–1420 (1971)], and Keefe and Benade [J. Acoust. Soc. Am. **74**, 320–332 (1983)]. At first sight, the results seem different. However, it can be shown that, at long wavelengths, they are essentially the same: The apparent density equals $\rho_B=\rho B^2/2[1-(1-B^2)^{1/2}]$, where ρ is the unperturbed air density and B is the ratio of the tube radius and the tore curvature radius. The transition of the torus into the regular cylindrical bore causes a correction which was measured by Keefe and Benade. This was verified by numerical calculations using conformal transformations. The effects of a torus on the tuning can be neutralized by reducing the bore in the torus, for sharp bends of up to some 5%.

4:45

1pMU9. Synchronized optical and acoustical measurements of trombone embouchure. Jay C. Bulen (Div. of Fine Arts, Baldwin 118, Northeast Missouri State Univ., Kirksville, MO 63501)

Outward striking- and inward striking-reed models have been proposed for representing brass players' lips [Sanoyesi *et al.*, Acustica **62**, 194–210 (1987)]. The models differ in the predicted relationship between mouthpiece pressure and lip displacement. To investigate this, Yoshikawa measured the phase relationship between mouthpiece pressure and lip strain as indicated by a strain gauge taped to the upper lip [J. Acoust. Soc. Am. **97**, 1929–1939 (1995)]. However, the relationship between strain and displacement have not been experimentally established, and Yoshikawa's assumed correspondence "is still a hypothesis which needs refinement" (p. 1931). Optical measurements are required. Synchronized optical and acoustical measurements of a trombonist's embouchure have been made under performance conditions, using an adaptation of techniques described in Sercarz *et al.* [Am. J. Otolaryngol. **13**, 40–44 (1992)]. Using strobed videoscopy, individual video fields are coordinated with mouthpiece pressure by means of timing signals. The phase relationship between mouthpiece pressure and lip displacement will be reported for a variety of fundamental frequencies and intensities. In addition, estimates will be presented of the aperture area and the mouthpiece volume swept out by the lips.

Session 1pSC

Speech Communication: Informal Workshop on Voice Perception II

David B. Pisoni, Cochair
Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, Indiana 47405

Jody Kreiman, Cochair
University of California, Los Angeles, Head/Neck Surgery, 31-24 Rehab, Los Angeles, California 90024-1794

The goal of this informal workshop, a continuation of Session 1aSC, is to bring together several researchers working on various aspects of voice perception. Historically, the study of voice has been treated as a more-or-less autonomous area quite distinct from other research problems in speech and hearing sciences. In this workshop, some of the traditional problems of voice classification and perception will be discussed and reviewed and then these efforts will be related to recent findings in speech perception and spoken word recognition which have shown important dependencies between traditional voice parameters and perceptual analysis of the speech signal. The contributors to the session include: Jody Kreiman, Diana VanLancker, Keith Johnson, Robert Remez, Peter Jusczyk, and David Pisoni. Please bring your slides and overheads and join in the discussion on some long-standing problems related to the "indexical" and "linguistic" functions of speech. For further information about this informal workshop, please feel free to contact D. B. Pisoni via E-mail: pisoni@indiana.edu. Everyone with an interest in speech perception, production, and spoken language processing is welcome to attend and contribute to the discussion of these problems.

MONDAY EVENING, 13 MAY 1996 MT. RUSHMORE, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Pitch, Periodicity and the Brain

Kenneth E. Gilbert, Chair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Pitch, periodicity, and the brain. William M. Hartmann (Phys. Dept., Michigan State Univ., East Lansing, MI 48824)

The perception of pitch forms the basis of musical melody and harmony. It is also the most precise of all our human senses. With imagination, this precision can be used experimentally to investigate the functioning of the auditory system. The Indianapolis tutorial will present auditory demonstrations from the zoo of pitch effects: pitch shifts, noise pitch, virtual pitch, dichotic pitch, and the pitches of things that are not there at all. The tutorial will introduce models of auditory processing, derived from contemporary psychoacoustics and auditory physiology, and will test these models against the experimental effects. It will conclude by describing the critical role played by pitch in the human ability to disentangle overlapping voices, as modeled by current work on artificial intelligence.

Session 2aAA

Architectural Acoustics: Research in Chair and Audience Absorption

Dana L. Kirkegaard, Chair
Kirkegaard Acoustics, Inc., 4927 Wallbank Avenue, Downers Grove, Illinois 60515

Chair's Introduction—8:30

Invited Papers

8:40

2aAA1. Sound absorption by seats in concert halls, occupied and unoccupied, and average sound absorption by the interior surfaces of halls. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138)

Average sound absorption coefficients were determined for the interior surfaces of concert halls based on measurements in five halls in which all finishes had been completed before installation of seats. Fifteen halls were selected for detailed analysis and divided into three groups with seats that are (1) lightly upholstered, (2) medium upholstered, and (3) heavily upholstered. Assuming relative humidites of 50% for unoccupied and 70% for occupied halls, "acoustical" areas (added 0.5-m edge effect) for seating blocks, and the above, average interior-surface, absorption coefficients, the sound absorption coefficients for occupied and unoccupied seats were calculated from reverberation measurements. Among the three groups, it is found that there are large differences in absorption coefficients for the seats at all frequencies, but particularly at low frequencies. The values are markedly different from those previously published [L. Beranek, J. Acoust. Soc. Am. **92**, 1–39 (1992)]. Where possible, photos and visual analyses of the chairs are made to compare with the data.

9:05

2aAA2. Sound absorption of occupied chairs as a function of chair design and audience clothing. Dana L. Kirkegaard (Kirkegaard Acoustics, 4927 Wallbank Ave., Downers Grove, IL 60515)

In 1989, a series of experiments was initiated at Riverbank Acoustical Laboratories to study the detailed relationships between chair design and audience absorption. Chairs were measured occupied and unoccupied. Many chairs were disassembled and tested as individual chair components to confirm specific correlations between design and data. Chair manufacturers responsed by redesigning chairs for further testing. Each series of tests was designed to minimize the number of experimental variables. An unupholstered reference chair provided a baseline for each series of occupied tests. The baseline measurements provide a means of comparing occupied chair measurements made with different occupants and/or changes of clothing. They also establish baseline absorption values for an audience in minimally absorptive chairs. Results of these occupied chair measurements show that differences in chair design are most highly correlated with data below 500 Hz, which ranged from 2 to 8 Sabins/person. Above 500 Hz, data ranged from 6 to 11 Sabins/person as a function of chair design and seasonal changes in occupants' clothing. Results support the hypothesis that chair design and seasonal clothing are critical in concert halls and other rooms with low residual absorption. [Research supported by Irwin Seating, JG Seating and Country Roads.]

9:30

2aAA3. An overview of the acoustical effects of an audience. J. S. Bradley (Acoust. Lab., Natl. Res. Council, Ottawa K1A 0R6, Canada)

This paper reviews the various effects of an audience on the acoustical properties of an auditorium. At the most basic level, the total sound absorption of both occupied and unoccupied chairs can be predicted in terms of the perimeter/area ratio of audience seating blocks. Using this technique, reverberation chamber measurements of occupied chairs can be used to estimate the reverberation times of fully occupied halls. This technique has been extended to provide estimates of typical sound-absorbing properties of occupied chairs for all sizes of seating blocks. Partially occupied seating areas present further problems. Audience effects on early- and late-arriving sound levels and early/late sound ratios are related to the geometry of the halls as well as to changes in the total sound absorption. Interaural cross correlations and lateral energy fractions are not greatly influenced by the presence of an audience. Changes to the more detailed spectra of the direct and early arriving sound show that the effect of the audience on the seat dip attenuation can vary with the type of seating as well as other properties of the hall.

9:55

2aAA4. Under-seat sound absorption for stadium/arena chairs. Robert C. Coffeen (Coffeen Fricke & Assoc., 14873 W. 9th St., Lenexa, KS 66215 and Programs in Architectural Eng., Marvin Hall, Univ. of Kansas, Lawrence, KS 66045)

For large audience spaces such as arenas and stadiums, it is often difficult to satisfactorily control the unoccupied reverberation period of the space because of the large volume in relation to the area of surfaces that are available for the practical installation of sound-absorbing material; and when chairs are of the plastic "stadium chair" type, it may be impossible to achieve a suitable stabilization of the reverberation period between unoccupied and occupied conditions. Of course, the use of chairs with sound absorbing upholstery is of major benefit in resolving these problems, but it is frequently found that "hard" chairs are selected for maintenance and cost reasons, and in spite of resulting acoustical conditions. However, this situation can be at least partially remedied by installing sound-absorbing material beneath each chair seat as an integral part of the chair construction. Laboratory sound absorption data, obtained in association with a major chair manufacturer, and field data obtained for a multipurpose arena indicate that this is a useful technique which can be applied when "hard" chairs are to be used for reasons not relating to acoustics.

10:20–10:35 Break

Contributed Papers

10:35

2aAA5. The resonator system of controlling the reverberation time in an existing hall. Shigeo Hase (Theatre Design Co., Ltd., 1-16-1 Shibata, Kita-ku, Osaka, 530 Japan) and Yoshihito Kobayashi (Asahi Glass Sound Proof System Co., Ltd., Tokyo, 110 Japan)

In this paper, the resonator system of controlling the reverberation time is examined in an existing hall with 500 seats and the room volume of 4160 m^3. The cavity of each chair with the volume of 0.012 m^3 was utilized as a resonator. Four holes of a 30-mm diameter in the lower board of chairs can be opened or closed by a sliding plate. By use of such a system of chairs, the reverberation time at the low-frequency range in the hall could be controlled as: at 100 Hz: 1.93–1.58 s; at 125 Hz: 1.69–1.44 s; at 160 Hz: 1.54–1.36 s, and at 200 Hz: 1.52–1.45 s.

10:50

2aAA6. Examination of a resonator system for sound power absorption by hall chairs in the low-frequency range. Yoshihito Kobayashi (Asahi Glass Sound Proof System Co., Ltd., 1-13-3 Ueno, Taito-ku, Tokyo, 110 Japan) and Shigeo Hase (Theatre Design Co., Ltd., Osaka, 530 Japan)

For the purpose of controlling reverberation time in the low-frequency range in a room, the variable system of sound absorption power by the chairs in the low-frequency range by the experiments in a scale model was examined. As the results of scale model experiments (1/10 scale) in the reverberation room, the absorption power was controlled in the low-frequency range by the opening and closing of holes of the resonator. Because of the achievement of the variable system of sound absorption of the actual chairs, an experiment on sound absorption power characteristics was made in the reverberation room of the chair with resonators given a variable with the quantity and the diameter of holes, a neck's length, and a cavity volume of the seat. The result was obtained that variable characteristics of the sound absorption power of the chairs with the resonator (four holes, 30 mm diameter of the hole, and neck length 10 mm) is 0.2 m^2 each for 125 or 250 Hz by changing the cavity volume of the seat in the experiment. The characteristics under the occupancy condition are nearly equal to it under nonoccupancy conditions.

11:05

2aAA7. Partially coupled chambers in auditorium design. Russell Johnson (Artec Consultants, Inc., 114 W. 26 St., 9 Fl., New York, NY 10001)

Our consulting group has completed 12 concert rooms or opera houses with controllable partially coupled chambers, and we have three either under construction or on the drafting boards. This paper provides a very brief overview of this approach to auditorium design and gives a rough idea of how this design feature might evolve during the next 5 years, with some reference to adjustability of the cubic volume of the primary chamber, and to the location of the openings into the secondary chamber.

11:20–11:50 PANEL DISCUSSION

TUESDAY MORNING, 14 MAY 1996 CELEBRATION B, 8:30 TO 11:05 A.M.

Session 2aEA

Engineering Acoustics, Noise and Structural Acoustics and Vibration: Noise Control in Internal Combustion Engines

Ahmet Selamet, Cochair
Department of Mechanical Engineering and The Center for Automotive Research, The Ohio State University, 206 West 18th Avenue, Columbus, Ohio 43210-1107

Raymond A. Kach, Cochair
Powertrain Operations, Ford Motor Company, 21500 Oakwood Boulevard, POEE Building, Dearborn, Michigan 48121

Chair's Introduction—8:30

Invited Papers

8:35

2aEA1. A new methodology of applying measured acceleration and forces or pressures as excitation on powertrain models for noise prediction. Nickolas Vlahopoulos, Richard D. Stark III, and Walter Kargus IV (Automated Anal. Corp., 2805 S. Industrial, Ste. 100, Ann Arbor, MI 48104-6767)

Numerical methods can be used to compute the vibration of powertrain systems or components and the corresponding radiated noise. Specifically, the finite-element method can be utilized in computing the structural vibration. The structural response will constitute the boundary conditions for the boundary-element method in computing the radiated noise. In this work a new methodology is presented, it allows a combination of measured accelerations, and forces or pressures, to apply as excitation. The innovative elements of this work are: (i) the methodology utilized in prescribing the acceleration at certain parts of the structure; and (ii) integrating the vibration analysis within the acoustics software and creating a single process for computing the noise. A generic six-cylinder engine model is used as an example to demonstrate the new capabilities. The structural finite-element model is utilized to compute the modal basis of the structure. Acceleration measured at the bearing locations, and pressure loads applied at the cylinder head (simulating combustion) comprise the excitation. The modal basis of the structure, the loads, the structural model, and the acoustic boundary-element model are utilized within the acoustics software in comparing the structural vibration and the corresponding radiated noise.

9:00

2aEA2. Sound-pressure simulations on a concentric tube resonator. Mehmet S. Ciray (Arvin North American Automotive, 1531 13th St., Columbus, IN 47201)

A concentric tube resonator was investigated to determine the accuracy of the numerical method used for sound-pressure simulations. The interaction between the tube and the cavity was through a perforated surface, of which the acoustical properties were experimentally determined [J. W. Sullivan, J. Acoust. Soc. Am. **76**, 479–484 (1984)]. These properties were imposed as boundary conditions to the finite-element analysis (FEA) code. Predicted and measured transmission losses were compared under no mean flow conditions. The work was then extended to include the effects of mean flow. Empirical equations given in literature were used to modify the acoustical properties of the perforated surface to incorporate the effects of flow [Garrison *et al.*, Pratt and Whitney Aircraft Rep. (1969)]. The mean flow terms in the scalar Helmholtz equation were treated explicitly in order to avoid the additional storage requirements for the FEA code. Special attention had been paid to the design of the FEA code, such that information can be seamlessly exchanged with a computational fluid dynamics (CFD) simulation, for possible future research.

9:25

2aEA3. Computational modeling of multiple-pass perforated tube silencers in the time domain. N. S. Dickey (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109-2121) and A. Selamet (Ohio State Univ., Columbus, OH 43210-1107)

To reduce the low-frequency radiated noise from automotive engines, intake and exhaust system silencers often employ perforated tube elements. These geometries commonly consist of one or more perforated tubes aligned axially within an external duct having a length significantly larger than any cross dimension. For such a silencer, the low-frequency acoustic performance may be analyzed by considering one-dimensional flow in each of the passages, coupled by the communication between regions through the perforate interface. The behavior of these configurations depends on numerous parameters, including porosity, orifice dimensions, mean and oscillating through flow and grazing flow levels, in addition to the overall geometry. In automotive applications, nonlinear orifice behavior may be expected. The present study applies a time domain fluid dynamic model to predict transmission loss of perforated tube elements. Since the Navier-Stokes equation is retained in nonlinear form, the technique shows promise for application to the breathing systems of firing engines, where numerous nonlinearities affect the validity of frequency domain analyses. Multiple-pass silencers with zero mean flow are modeled within an extended impedance tube configuration. The effects of overall geometry, porosity, and sound-pressure level are studied and limitations of the one-dimensional assumption are addressed. [Work supported by Ford Motor Co.]

9:50–10:05 Break

Contributed Papers

10:05

2aEA4. Predictive acoustic modeling applied to the control of intake/exhaust noise of internal combustion engines. Peter O. A. L. Davies (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO17 1BJ, UK)

The application of validated acoustic models to intake/exhaust system acoustic design is described with reference to a sequence of specific practical examples. These include large turbocharged diesel generating sets, truck engines, and high-performance petrol engines.

10:20

2aEA5. Side branches in automotive intake systems as noise generators. A. Selamet (Dept. of Mech. Eng. and Ctr. for Automotive Res., Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210-1107) and P. M. Radavich (Univ. of Michigan, Ann Arbor, MI 48109-2121)

It is well established that the coupling of flow induced vortices and acoustic resonances in a closed side branch can lead to the production of both large wave amplitudes in this duct and excessive noise levels in the main duct. The present study concentrates on the propagation of noise in an automotive intake system induced by the closed side branch of an exhaust gas recirculation tube. The effects of (1) altering the side branch length, and (2) the addition of an area expansion in the side branch are investigated to modify the acoustic resonance characteristics. [Work supported by Ford Motor Co.]

10:35

2aEA6. Internal combustion engine noise analysis and production using a novel multipulse excited time series modeling technique. Scott A. Amman (Ford Motor Co., 3003 Prairie, Royal Oak, MI 48073) and Manohar Das (Oakland Univ., Rochester, MI 48309-4401)

Automobile manufacturers have learned that customers have well-defined notions as to what an automobile engine should sound like for the various vehicle segments (luxury, sportscar, economy, etc.). Therefore, it is advantageous to model and synthesize various types of engine sounds for customer evaluation. Much work has been done in the area of speech modeling and synthesis based on physical assumptions of human speech production. The intent of this paper is twofold. First discussed are the similarities and differences between speech and engine noise modeling and synthesis problems. Second, application of techniques learned for speech synthesis to that of automotive noise production is demonstrated. Early models of speech production used periodic pulse excitation in a linear predictive coding (LPC) framework. This technique is used to synthesize engine noise. It is then compared to that of a new suboptimal multipulse method proposed by the authors. It is hoped that this paper will pave the path for a better understanding of the relationship between the two systems so that much of the work performed in speech analysis and production can be applied to the analysis and synthesis of automotive internal combustion engine sounds.

10:50

2aEA7. Active control of sound radiated from an open-ended duct using sound intensity. S.-W. Kang and Y.-H. Kim (Ctr. for NOVIC, Dept. of Mech. Eng., KAIST, Science Town, Taejon 305-701, Korea)

Sound intensity based active control for the reduction of sound radiated out of the duct exit is studied experimentally as well as theoretically. The active intensity control strategy is derived based on the relation of the exterior sound field out of the duct exit and interior sound field of the duct. One of the features of the active intensity control strategy is that the control performance can be maintained regardless of the sensor location, compared with the conventional local sound-pressure control methods at either interior downstream or exterior field position. Simple numerical examples are demonstrated to compare the acoustic characteristics of the active intensity control scheme with the other control strategies. For a more practical comparison with the conventional sound-pressure control strategy, an experiment, based on a time-domain adaptive filtering with an intensity sensor filtered-x LMP (least-mean-product) algorithm, is performed. From the experimental results, it is shown that the exterior sound field is much more observable by sensing the active intensity than by sensing the sound pressure. It is also demonstrated that the active intensity control performances are superior to the sound-pressure control ones.

TUESDAY MORNING, 14 MAY 1996 HARRISON'S, 8:00 TO 11:50 A.M.

Session 2aMU

Musical Acoustics: Reed Instruments: Research and Performance

James M. Pyne, Chair

School of Music and Center for Cognitive Science, The Ohio State University, 1866 College Road, Columbus, Ohio 43210

Invited Papers

8:00

2aMU1. Acoustics of American reed organs. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

The instrument that has come to be called the "American" reed organ shares with its relatives the accordion, concertina, and harmonica the use of free reeds, but differs from its reed organ cousin, the harmonium, in the use of a vacuum exhauster rather than a pressure bellows to drive the reed vibration. The acoustical properties of the instrument are summarized along with the results of some recent experimental studies. These investigations have explored variations in properties of both the reed vibration and the resulting sound as a function of pressure. Effects of airflow rate, and reed cell dimensions have also been studied. Both live and recorded musical examples are provided.

8:30

2aMU2. Music for the reed organ. Edward A. Peterson (Reed Organ Society Bulletin, 101 Ulen Blvd., Country Club Park, Lebanon, IN 46052-3202)

Members of the American Guild of Organists and the Reed Organ Society will perform on instruments from the Indianapolis area. Selections will be presented which illustrate the musical range of the reed organ. Commentary on the principles of operation of the instrument and playing techniques will be provided.

9:00

2aMU3. "Reed all about it:" The giant grass that provides reed cane for musical instruments. Marcus Eley (Rico Intl., 8484 San Fernando Rd., Sun Valley, CA 91352)

Cane reeds for musical instruments are made from the stalks of arundo donax, a giant grass. Although arundo donax grows wild througout the world only certain conditions of soil and climate will produce cane that is suitable for music making. The controlled conditions of plantation growth, as in the plantation system used by Rico International, are proving to be successful in growing cane that is more consistently of high quality. Nurturing and careful harvesting bring the mature stalks to the stage where they can be carefully sun-dried in racks for many months and finally cut, between nodes, into tubes. The cane tubes are then shipped to Sun Valley for the manufacture of musical instrument reeds, a process that involves special tooling and unique manufacturing techniques.

9:30

2aMU4. Techniques for defining clarinet reed quality via computerized light transmission analysis. James M. Payne (School of Music and Ctr. for Cognit. Sci., Ohio State Univ., 1866 College Rd., Columbus, OH 43210)

A basic and well-recognized problem in clarinet and saxophone (single-reed) performance is maintaining consistency in high-quality generation and manipulation of tones. This generative process, involving reed, mouthpiece, and the player's embouchure, produces the instrumental "voice" of the performer and is fundamental to artistic communication. Playing skill and a well-designed mouthpiece cannot overcome the liability of poorly vibrating arundo donax, the natural material used in reed manufacture. Inconsistency in quality leads professionals to spend as much as $2500 per year on reeds and to perform on relatively few of those purchased. Highly skilled performers agree that the identification of fine performance reeds cannot be accomplished by visual inspection. Trial by playing is the customary means of determining quality and one must purchase reeds before these trials can be made. Even these carefully selected reeds degrade and become unplayable after approximately 10–20 h of performance. Computerized light transmission analysis (TRACOR) has proven to be of value in determining performance characteristics of clarinet reeds.

10:00–10:15 Break

Contributed Papers

10:15

2aMU5. Observed vibration patterns of clarinet reeds. P. L. Hoekje and G. Matthew Roberts (Dept. of Phys., Univ. of N. Iowa, Cedar Falls, IA 50614-0150)

It has been shown that a resonance of the reed contributes significantly to the playing behavior of the single-reed woodwind [S. C. Thompson, J. Acoust. Soc. Am. **66**, 1299–1307 (1979)]. As an extended object, the reed is actually expected to possess many resonances, and these have been predicted in FEA calculations [D. Casadonte, J. Acoust. Soc. Am. **94**, 1807(A) (1993)] and observed in vibration spectra of isolated reeds [D. H. Keefe and S. Waeffler, J. Acoust. Soc. Am. **94**, 1833–1834(A) (1993)]. The current study used holographic interferometry to identify the vibration patterns of a clarinet reed mounted on a mouthpiece, as driven by an acoustic field internal to the mouthpiece. Without a player's lips to shorten the vibrating section, the lowest frequency and most easily driven resonance is in the region 1500 to 2000 Hz and corresponds to the lowest mode of a rod clamped at one end. The second and third resonances are near 4000 Hz. The second mode displays twisting motion, but with a surprising degree of asymmetry, while the third mode is similar to the second mode of a rod clamped at one end.

10:30

2aMU6. Effect of moisture on flexural wave fundamental frequency of clamped-end cane reeds. Jacques R. Chamuel (Sonoquest Adv. Ultrason. Res., P.O. Box 81153, Wellesley Hills, MA 02181-0001)

Experimental results are presented demonstrating the effect of moisture on the fundamental flexural wave frequency of clamped-end cane reeds. The flexural wave velocity decreases as the moisture level increases. The change in flexural wave velocity seems to be dominated by softening of the cane rather than by the added water mass. Plots are presented comparing the dependence of flexural wave fundamental frequency on moisture for different types of cane reeds. The results indicate that the fundamental frequency of a cane reed may decrease by as much as 30% as it becomes saturated with water. The flexural rigidity diminishes when the cane is moistened. The findings are important for the characterization and evaluation of reeds where hidden uncontrolled trapped moisture significantly affect the response, vibration, and stiffness of the reed.

10:45

2aMU7. Handmaking clarinet reeds: New tools accelerate the process. Timothy Anderson (School of Music, Ohio State Univ., 1866 College Rd., Columbus, OH 43210)

The vast majority of clarinetists worldwide (professionals, students, and amateurs) perform on commercially manufactured reeds. However, a number of performers, including many professionals, maintain that commercial reeds can never match the high quality of reeds handmade from tube by the performer. For clarinetists this time-consuming practice was once standard procedure and still is with double-reed performers (oboe and bassoon). This reed-maker supports the position that the best reeds are produced by the performer. Recently, tools have become available that accelerate the process of making one's own clarinet reeds. Using some of these tools a reed will be fashioned from arundo donax tube during the presentation and its sound will be demonstrated.

11:00–11:15 Break

Invited Paper

11:05

2aMU8. *I Clarinetti Virtuosi:* Chamber ensemble of the Ohio State University. James Pyne and Elisabeth Stimpert (School of Music, Ohio State Univ., 1866 College Rd., Columbus, OH 43210)

The clarinet ensemble continues to emerge as a capable vehicle for art performance. It can, more than any other family of wind or brass instruments, closely reproduce the range, flexibility, and homogeneity of the string ensemble. These characteristics will be briefly discussed and demonstrated using the individual instruments of the clarinet family from contra-bass to piccolo clarinet including basset horn, a favorite instrument of Mozart. *I Clarinetti Virtuosi* are clarinet performance majors at The Ohio State University, graduate students and undergraduate seniors. They will perform a variety of works that display the wide range of musical and virtuosic capability of the ensemble.

Session 2aNS

Noise: Hearing Conservation Issues from a Global Perspective

Larry H. Royster, Chair
Mechanical and Aerospace Engineering Department, North Carolina State University, Raleigh, North Carolina 27695-7910

Chair's Introduction—8:30

Invited Papers

8:35

2aNS1. Noise and hearing conservation regulations throughout the world. Alice H. Suter (Alice Suter and Assoc., 575 Dogwood Way, Ashland, OR 97520)

The advent of electronic communication enables professionals in noise and hearing conservation to communicate more effectively than ever before with their colleagues in other nations. This fact, added to the harmonization of noise standards by the European community, enables people to watch the development of trends throughout the world. The trend in nations' noise standards is toward the 3-dB (equal-energy) rule, toward A-weighted permissible exposure limits below 90 dB, and toward more specific requirements for audiometric testing and the wearing of hearing protection devices. In addition to protecting workers against hearing loss, several nations include requirements to protect against other adverse effects of noise. Some nations have issued noise standards for specific workplaces, processes, and equipment. Enforcement of noise standards varies considerably among nations, with some nations viewing noise standards merely as recommendations, and others enforcing them seriously.

9:00

2aNS2. Various international approaches for evaluating impulse noise. Daniel L. Johnson (Biophys. Operation, EG&G MSI, 2450 Alamo SE, P.O. Box 9100, Albuquerque, NM 87119)

For impulses with a peak sound-pressure level (SPL) under 140 dB, ISO1999 provides a standardized approach using total A-weighted energy. Some countries may use a small correction factor in addition. At the present time, most approaches for evaluating impulse noise above 140 dB use the peak sound-pressure level and some estimate of duration of the waveform. There is no recognized standard method for evaluating such impulses. Furthermore, the problem with this peak and duration method is that some waveforms with the same peak and duration parameters are far more hazardous than others. As will be discussed, each major country is using a slightly different approach. However, there is a NATO working group investigating a common approach. A part of this approach may be to use some type of weighted energy.

9:25

2aNS3. International activities in the use, standardization, and regulation of hearing protection. Elliott H. Berger (E-A-R/ Cabot Safety Corp., 7911 Zionsville Rd., Indianapolis, IN 46268-1657)

Prevention of hearing loss due to noise exposure often means one thing—the consistent and correct use of hearing protection. Since noise-induced hearing loss has been identified as one of the leading occupational diseases and injuries, there are millions of workers worldwide in need of, or already currently wearing, hearing protection. How well they utilize the products they are provided, as well as the extent and quality of government enforcement activities, is influenced, in part, by the standards and regulations that govern the testing and labeling of such products. As such, this paper will summarize current international standards on testing, labeling, and performance of hearing protectors as well as reviewing field performance and anecdotal observations pertaining to actual use. Attention will focus on ISO, CEN (Community on European Normalization), and Australian activities, as well as on a new draft ANSI standard (S12.6-199X) pertaining to the laboratory measurement of the real-ear attenuation of hearing protectors, and on the efforts of the National Hearing Conservation Association Task Force on hearing protector effectiveness to craft recommendations for the labeling of hearing protection devices sold in the U.S.

9:50

2aNS4. The audiometric phase of the hearing conservation program: Summary of activities from a global standpoint. Larry H. Royster (MAE Dept., N. C. State Univ., Raleigh, NC 27695-7910) and Julia Doswell Royster (Environ. Noise Consultants, Raleigh, NC 27622-0698)

As part of the activities of ISO/TC 43/WG5: Evaluating the effectiveness of hearing conservation programs, a questionnaire was mailed out to the interested International Standards Organizations membership requesting information related to hearing conservation program activities. Part of this questionnaire dealt with the audiometric activities in each country. This paper will summarize the findings from this survey and discuss the past and present activities of ISO/TC 43/WG5. The significant differences that exist in the attitude toward obtaining and making use of audiometric data around the world will be discussed.

Contributed Papers

10:30

2aNS5. Repeatability and reproducibility in hearing protector testing. John R. Franks, William J. Murphy (Bioacoust. and Occup. Vib. Sec., Natl. Inst. for Occup. Safety and Health, MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998), and Stephen D. Simon (Natl. Inst. for Occup. Safety and Health, Cincinnati, OH 45226-1998)

Earlier research [Royster *et al.*, J. Acoust. Soc. Am. **99**, 1506–1526 (1996)] has reported on a new laboratory test method to predict the amount of attenuation hearing protectors provide to workers in the real world. The subject fit method (SF) employs the use of subjects who are audiometrically competent but naive with respect to hearing protector use and who must rely only upon the manufacturer's instructions to fit a protector for testing. One unresolved issue has been the number of subjects and number of repeated tests per subject necessary to achieve the desired levels of repeatability (test–retest within a laboratory) and reproducibility (test–retest between laboratories). Attenuation data from four laboratories collected with the same protocol showed that estimates of variance for the reproducibility across hearing protectors (Bilsom Muff, E-A-R plug, EP100 plug, V-51R plug) for informed user fit (IUF) were greater than estimates of the SF variance. The estimates of variance for the repeatability across protectors and frequency were comparable between the IUF and SF conditions. The average IUF attenuations across protectors and frequency were greater than the average SF attenuations for all of the plugs.

10:45

2aNS6. Hearing hazard from the noise of air bag deployment. G. Richard Price (Human Res. and Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD 21005-5425), Stephen W. Rouhana (GM Res. Lab., Warren, MI 48090-9055), and Joel T. Kalb (Army Res. Lab., Aberdeen Proving Ground, MD 21005-5425)

Recent studies have questioned the adequacy of current noise standards for intense impulses, especially those with large low-frequency content, a category which includes air bag deployments. Unfortunately, almost no tests have been done with real ears and air bag impulses. Such data are also needed for development of theory and validation of our hearing loss model [G. R. Price and J. T. Kalb, J. Acoust. Soc. Am. **90**, 219–227 (1991)]. Therefore, 32 anesthetized cats positioned at the driver and passenger locations in a pickup truck were exposed in pairs to one air bag deployment (electrically initiated). Hearing was tested at 1, 2, 4, 8, and 16 kHz by evoked-response audiometry before exposure, immediately after, 1 month, and 6 months later. Exposure conditions included doors open, compartment closed, and closed compartment sealed with tape; seven exposures to passenger bag only and nine to driver and passenger bags. Pressures at inboard ears ranged from 167- to 173-dB peak and B durations from approximately 50–150 ms, with unweighted energies as high as 4000 J/m^2 (or 8 h $L_{\rm EQA}$=95.5 dB). All ears showed significant hearing losses immediately, averaging 60-dB threshold shift at 4.0 kHz which resolved to an average PTS of 37 dB.

11:00

2aNS7. Modeling auditory hazard from impulses with large low-frequency components. G. Richard Price and Joel T. Kalb (Human Res. and Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD 21005-5425)

In order to validate the model of hearing hazard from intense sounds [G. R. Price and J. T. Kalb, J. Acoust. Soc. Am. **90**, 219–227 (1991)], hearing loss data have been sought from impulse noise exposures which extend the range of exposures already examined with the model. One such impulse (reported in Price *et al.* at this meeting) is that produced by air bag deployment, which contains a mixture of high-frequency energy often superimposed on a low-frequency pedestal, reaching peak pressures near 170 dB and with B durations of 150 ms. The model's predictions for firearms impulses in the same intensity region correlated well with actual threshold shifts [Price *et al.*, J. Acoust. Soc. Am. **97**, 3343 (1995)]; however, the hazards calculated with the air bag impulses were generally too large. Examination of the data suggested that middle-ear muscle activity needs to be accounted for, that corrections for the angle of incidence of the wavefront are needed, and that the existing model included too much damping for large displacements of the middle ear.

11:15

2aNS8. Maximum safe exposure levels for intense reverberant impulses in an enclosure. James H. Patterson, Jr. (202 Cedar Chase, Dothan, AL 36303), Daniel L. Johnson, and John T. Yelverton (EG&G MSI, Albuquerque, NM 87119)

Studies were undertaken to determine the maximum safe-exposure levels in a reverberant wave environment like that produced by firing an antiarmor weapon from a small room. The approach was first, to establish a threshold of injury for organs other than the ear then to determine whether these levels were safe for the ear. Using sheep as an animal model, the threshold of nonauditory injury was found to be approximately 190 dB peak SPL (65 kPa) with a B duration longer than 200 ms for one impulse and approximately 187 dB peak SPL (46 kPa) for three impulses. Two groups of 40 animals were used to establish statistical confidence in the "no injury levels." The auditory effects were investigated using human volunteers exposed to a progression of levels from 168 to 185 dB at the ear for one impulse and then, to two and three impulses at 183 dB. A temporary threshold shift (TTS), determined 2–4 min post exposure, was used as an indicator of auditory effect. The volunteers wore an earmuff modified to simulate a poor fit, during the exposures. No significant TTS in 59 volunteers was observed, indicating no effects on hearing up to the maximum safe-exposure levels for other organs.

Session 2aPAa

Physical Acoustics and Bioresponse to Vibration and to Ultrasound: Workshop on Therapeutic Applications of Medical Ultrasound I

Floyd Dunn, Chair
Bioacoustics Research Laboratory, University of Illinois, 405 N. Mathews Avenue, Urbana, Illinois 61801

This Workshop is an attempt to bring a "small meeting" environment into a big meeting. In addition to the tutorial and special presentations outlined below and in the later two sessions (2pPAa, 3aPAa), there will be significant participation by other workshop attendees; discussion among participants is a major objective of this program. Therefore, a specific timetable will not be followed during the workshop.

Chair's Introduction—8:30

Invited Papers

2aPAa1. Remotely induced shear acoustic waves in tissues by radiation force of focused ultrasound. Armen P. Sarvazyan (Dept. of Chemistry, Rutgers Univ., Piscataway, NJ 08855-0939) and Oleg V. Rudenko (Moscow State Univ., Moscow, Russia 119899)

A new acoustic approach to medical imaging and diagnostics based on the remote evaluation of shear elasticity modulus of soft tissues is considered. The new method, called shear wave elasticity imaging, is based on the detection of shear waves remotely generated in tissues by radiation pressure of amplitude-modulated focused ultrasound. Remotely induced shear wave is fully attenuated within a short distance and the induced strain in the tissue can be extremely localized. By choosing appropriate temporal characteristics of the amplitude-modulated ultrasonic pulse, the volume of tissue involved in the mechanical excitation can be kept on the order of $1=A0$ cm^3, in contrast to other methods of elasticity imaging where the complete organ is subjected to shear stress. Consequently, evaluation of viscoelastic properties of tissue is greatly simplified since trivial boundary conditions can be assumed and an infinite medium model can be used to reconstruct the mechanical properties of tissue. Analytical equations describing the radiation force field induced by nonlinear focused ultrasound are derived and propagation of shear waves generated in tissues is considered. A possibility of deliberately exploiting the nonlinear effects to enhance the radiation pressure in a small area near the focus of the beam is shown.

2aPAa2. Acoustic nonlinearity and radiation pressure in ultrasonic bioeffects. Oleg V. Rudenko (Phys. Dept., Moscow State Univ., Moscow, Russia 119899) and Armen P. Sarvazyan (Rutgers Univ., Piscataway, NJ 08855-0939)

Most of the biomedical applications of ultrasound are based on the use of focused ultrasound with a high concentration of energy in the focal region and, respectively, with significant contribution of nonlinear phenomena in ultrasonic bioeffects. The main objective of the current paper is to provide a clear understanding of basic phenomena in nonlinear acoustic fields of focused ultrasound, and to derive simplified equations that would enable physicists and engineers to optimize parameters of biomedical devices for particular applications. Special attention is paid to the contribution of acoustic nonlinearity to the second-order quantities, such as radiation pressure and temperature rise in ultrasonic fields in tissues. Based on asymptotic methods recently developed in nonlinear acoustics, analytical solutions of the equations for the radiation force induced in a dissipative medium are considered. The relationships between parameters of acoustic field and characteristics of biological tissues are quantitatively analyzed. Possible roles of nonlinear acoustic phenomena and ultrasound radiation force in various features of interaction of ultrasonic waves with biological media are discussed.

2aPAa3. Finite amplitude distortion in derating of ultrasound fields. T. Christopher and E. L. Carstensen (Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

All ultrasound exposure indices that have been formulated up to the present time assume that the propagation of ultrasound is linear. In fact, under most exposure conditions for which biological effects may be a concern, sound propagation is highly nonlinear. A nonlinear propagation model has been used to evaluate the nature of the effects that occur under realistic exposure conditions encountered in diagnostic procedures. Because of the way that the thermal index is defined, it turns out that ignoring nonlinear propagation leads to underestimates of tissue temperature increments that typically are less than 40%. As currently implemented, the mechanical index may be underestimated by more than a factor of 2 because it ignores the saturation of the sound fields that result from nonlinear propagation. For large propagation distances in soft tissues (e.g., 10 cm at 3 MHz in liver), however, it is physically difficult to exceed tissue pressures corresponding to MI>2 because of these same saturation phenomena.

2aPAa4. Lung hemorrhage from exposure to ultrasound. Diane Dalecki, Sally Z. Child, Carol H. Raeman, David P. Penney, and Edwin L. Carstensen (Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Thresholds for lung hemorrhage from exposure to pulsed ultrasound have been determined in neonatal mice (24–36 h), juvenile mice (14 days), and adult mice (8–10 weeks), and neonatal (24–36 h) and young (10 days) swine. The threshold at 2 MHz is approximately 1 MPa and the threshold increases gradually with frequency in the range from 0.1 to 4 MHz. Positive and negative pressure pulses are equally damaging. Following exposure, suprathreshold lesions are not progressive with time and are repaired by usual physiological mechanisms. Although the unique sensitivity of lung tissue to ultrasound is associated with the presence of air in the alveolae, this does not necessarily implicate acoustic cavitation as the responsible mechanism. Thin sections of murine lung are more sensitive than thicker regions probably because of reduced reflection at the surface of the lung tissue.

2aPAa5. Effects of ultrasound on binding of tissue plasminogen activator to fibrin. Farhan Sidiqi, Julie V. Braaten, and Charles W. Francis (Hematology Unit, Dept. of Medicine, and Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, 601 Elmwood Ave., Rochester, NY 14642)

To investigate the mechanism of ultrasound-enhanced fibrinolysis, the effect of ultrasound on binding of tissue plasminogen activator (t–PA) to fibrin was examined because t–PA–fibrin interactions are important in determining the rate of fibrinolysis. A novel system to quantitate t–PA–fibrin interactions was developed in which radiolabeled, active site blocked t–PA flowed through a fibrin gel at constant rate and temperature, and specific binding was determined by monitoring incorporation of radiolabel. Exposure to 1-MHz ultrasound to 2 W cm^2 resulted in no significant change in the kD of t–PA binding to either cross-linked or noncross-linked fibrin. Ultrasound, however, significantly increased the B_{max}, and the molar binding ratio increased in the presence of ultrasound from 1.85 (mol t–PA/mol fibrin) to 2.54 (mol t–PA/mol fibrin) for noncross-linked fibrin and from 1.19 (mol t–PA/mol fibrin) to 1.43 (mol t–PA/mol fibrin) for cross-linked fibrin. Ultrasound resulted in no significant changes in binding of t–PA to fibrin monomer in the same system. It is concluded that exposure to ultrasound increases binding of t–PA to polymerized fibrin, possibly by exposing additional binding sites in the polymer which are inaccessible in the absence of ultrasound.

2aPAa6. Acceleration of thrombolysis by ultrasound in two rabbit models. Charles W. Francis and Patrick N. Riggs (Hematology Unit, Depts. of Medicine and Vascular Surgery, Univ. of Rochester, 601 Elmwood Ave., Rochester, NY 14642)

Ultrasound accelerates fibrinolysis *in vitro* at intensities that are potentially applicable clinically to enhance thrombolytic therapy. To extend these findings *in vivo*, the effects of ultrasound on fibrinolysis induced by streptokinase have been examined in two rabbit models. In a rabbit model of fibrinolysis in small vessels, scalpel cuts were made in the ear and rabbits were then rested 2 h to allow maturation of hemostatic plugs. Bleeding occurred when clots in small ear vessels were lysed during thrombolytic therapy. In a prospective study, administration of streptokinase alone results in bleeding after 18 ± 3 min (mead ± SEM), but addition of ultrasound (1 MHz, 1 W cm^{-2} and a 50% duty cycle) shortened the time to bleeding to 7 ± 4 min ($p<0.05$). The effect of ultrasound was also examined in a rabbit arterial thrombolysis model and, thrombolysis occurred in 9/17 (53%) of animals receiving both streptokinase and ultrasound, and this was significantly greater than the rate in animals receiving streptokinase alone (2/15, 13%, $p=0.025$). Light and electron microscopic examination revealed minor changes in platelet accumulation on the thrombus and minor changes in the vessel intima of ultrasound-exposed vessels. These findings indicate that externally applied ultrasound accelerates fibrinolysis *in vivo*.

2aPAa7. Ultrasound increases fluid permeation in fibrin gels and reversibly disaggregates large fibers. Julie V. Braaten, Farhan Sidiqi, Edwin L. Carstensen, and Charles W. Francis (Hematology Unit, Dept. of Medicine, and Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, 601 Elmwood Ave., Rochester, NY 14642)

Since ultrasound accelerates fibrinolysis and since transport of enzymes into clots is an important determinant of the rate of fibrinolysis, the effect of ultrasound on fluid permeation through fibrin gels *in vitro* was examined. Gels of purified fibrin were prepared in plastic tubes, and the rate of pressure-mediated fluid permeation was measured. Exposure to 1 MHz ultrasound at 2 W cm^{-2} and duty cycle of 5 ms on, 5 ms off resulted in a significant ($p=0.005$) increase in flow through the gel of $29.0 \pm 4.2\%$ (mean ± SEM). The increase in flow was intensity dependent and not due to heating or fragmentation but was reduced by degassing, suggesting a role for cavitation. Scanning electron microscopy was used to examine potential effects of ultrasound on fibrin structure, which is the primary determinant of flow resistance. Fibrin gels were fixed and prepared for microscopy before, during, and after exposure to 1-MHz ultrasound at intensities from 4 to 8 W cm^{-2}. Ultrasound caused a significant ($p<0.000001$) increase in fiber density and decrease in fiber diameter which was reversible when ultrasound was switched off. These results indicate that ultrasound exposure can induce a significant reversible change in fibrin structure and fluid permeation that could contribute to the ultrasonic enhancement of fibrinolysis.

Session 2aPAb

Physical Acoustics: Materials Characterization

John A. Burkhardt, Chair
Department of Engineering, Indiana-Purdue University at Ft. Wayne, 2101 East Coliseum Boulevard, Ft. Wayne, Indiana 46805-1499

Contributed Papers

8:30

2aPAb1. Determination of velocity and attenuation of shear waves using broadband pulse technique. Junru Wu (Dept. of Phys. and Mater. Sci. Program, Univ. of Vermont, Burlington, VT 05405)

The ultrasonic spectroscopy (broadband pulse) technique was applied to simultaneously measure phase velocity and attenuation coefficient of shear waves in a solid in the megahertz frequencies. This technique is an extension of the ultrasonic spectroscopy technique currently used in determining dispersion of longitudinal waves. Sources of error in measurements including diffraction loss and nonlinear distortion will be discussed. [Work supported by Hewlett–Packard Co.]

8:45

2aPAb2. Quality characterization of an electrical solder point. Mohamed Ezzaïdi, Ali Moudden (Lab. d'Instrumentation et de Mesures, Faculté des Sciences, Univ. Ibnou Zohr, Agadir, Morocco), Dominique Decultot, and Gérard Maze (Université du Havre, 76610 Le Havre, France)

The Lamb waves are guided modes which propagate in a plane layer. The A_i antisymmetric and S_i symmetric ($i=1,2,\ldots$) Lamb waves have an infinite phase velocity in low frequencies (cutoff frequencies), and the phase velocity limit, in high frequencies, is equal to the velocity of the shear wave in the isotropic solid material. The thickness resonance of an elastic plate normally insonified by an ultrasonic plane wave is related to the cutoff frequencies. From these resonance frequencies, the plate thickness can be easily deduced. In this presentation, a focused ultrasonic transducer with a large bandpass to characterize an electrical solder point between two steel sheets is used. This solder is abundantly used to manufacture cars. An experimental ultrasonic pulse method (pulse-MIR) allows us to plot acoustic spectra. The solder is placed in the focal spot of the beam. The transducer can be moved in a parallel direction to the surface of the sheet steel. Many time signals are recorded along a diametrical line with a small step. For each time signal a spectrum is computed. The results are shown with a graphical software. The false color images allow us to evaluate the solder quality.

9:00

2aPAb3. Experimental study of the Scholte wave interaction with a square section defect. Alain Tinel, Hugues Duflo, and Jean Duclos (Lab. d'Acoust. Ultrason. et d'Electron., Univ. du Havre, Place Robert Schuman, 76610 Le Havre, France)

In this work, an experimental study of surface wave diffraction on a substrate immersed in water is presented. The Scholte wave was generated by bulk wave conversion at the extremity of the substrate. The sample was a plane plate of duralumin 10 mm thick and 200 mm long which had a defect (a tenon or a mortise) with square section dimensions close to the wavelength of the wave in the fluid. Only one obstacle perpendicular to the direction of propagation of the incident wave was investigated. The acoustic pressure at 300 mm around the diffraction zone was recorded. Phenomena of mode conversions were observed. They were the same as in the Scholte wave diffraction by a dihedral: the generation of the transmitted and reflected Scholte waves, and the generalized Rayleigh waves. The study was extended to the A wave diffraction on a thin plane plate. The dimensions of the square section of the defect were half of the plate thickness. In this case, the waves generated were generalized Lamb waves.

9:15

2aPAb4. Analysis and optimization of wedge transducers for Lamb wave excitation. F. Levent Degertekin and Butrus T. Khuri-Yakub (Stanford Univ., E. L. Ginzton Lab., Stanford, CA 94305)

Wedge transducers are commonly used to excite and detect Lamb waves in solid plates for NDE purposes. In this paper, the analysis and design of wedge transducers for efficient, single mode Lamb wave excitation in general multilayered anisotropic plates is presented. The incident acoustic field distribution on the surface of the plate due to a wedge transducer with arbitrary apodization is calculated using the angular spectrum method. Then, the particle velocity and stress field distributions in the plate in response to this source is obtained by the surface impedance approach. This field distribution can be expressed as a sum of Lamb wave modes in the structure, since the Lamb waves form an orthogonal mode basis. Once the mode amplitudes are found, the excitation efficiency for the desired Lamb wave mode is calculated. The results indicate a trade-off between efficiency and multimode excitation, as expected. To achieve both high efficiency and single mode excitation, an optimization scheme is used where the parameters involve the frequency of operation, dimension of the wedge, and the apodization function. The results of this analysis are presented for various practical examples.

9:30

2aPAb5. Experimental study of Lamb waves having a negative energy velocity in a thin plane plate immersed in liquid. Yannick Eudeline, Hugues Duflo, and Jean Duclos (Lab. d'Acoust. Ultrason. et d'Electron., Univ. Le Havre, 76610 Le Havre, France)

Lamb waves having a negative energy velocity have been theoretically predicted; they are called negative waves for short. This word is used if the phase velocity is in the opposite direction to the energy velocity. When a plate is immersed in a liquid, this phenomenon still occurs. These waves become leaky as their energy is being reradiated into the liquid. Any negative wave has the same limit frequency as a Lamb wave, and the same label with a prime (′) sign (i.e., S_2-S_2'). Some common solids were studied: duralumin, stainless steel, brass, and glass. The negative waves were generated (as Lamb waves are) by a tone burst bulk wave and observed after propagation over the plate by translating one of the transducers in order to receive the negative waves. The existence of several negative waves was proved; especially the A_3' glass wave which can be observed quite easily. When the frequency $\times$ thickness product decreases, several properties were observed in accordance with the theory: (i) the absolute value of energy velocity increased and reached a maximum near the cutoff frequency, (ii) the attenuation increased. A time-frequency study follows which enables the calculation of the energy velocity.

9:45

2aPAb6. Experimental study of *A* wave propagation along a plane or curved plate in contact with two different liquids. Loïc Martinez, Jean Duclos, Alain Tinel, and Naum Veksler (Lab. d'Acoust. Ultrason. et d'Electron., Univ. Le Havre, Place Robert Schuman, 76610 Le Havre, France)

The *A* wave propagation was first studied on 200-mm-long plane plates made of brass. The plates were 0.2 or 0.1 mm thick. The external fluid was always distilled water, and air, water, propanol, or glycol was used as internal fluids. With propanol or glycol, a second wave was observed, called A^*, similar to the *A* wave but having a strong attenuation in the direction of propagation. Using time frequency techniques, group dispersion curves were shown to be a function of the frequency thickness product in the range 0.1–1 MHz mm. When possible, the phase velocity was derived from experimental results using the phase of the signal spectrum. Experimental results presented a very good agreement with theoretical values. The *A* and A^* wave propagations were then investigated using cylindrical brass boxes. The thickness was also 0.2 or 0.1 mm and the radius was 60 mm. The box was filled with air, distilled water, propanol, or glycol. The experimental dispersion curves were comparable to those in the case of plane plates. If propanol or glycol were used, A^* or *A* waves (respectively) were detected tangentially. Their attenuation was also measured.

10:00–10:15 Break

10:15

2aPAb7. Elastic interfacial waves in orthotropic interlayers. Dimitrios A. Sotiropoulos (Dept. of Mech., Mater. and Aerospace Eng., Illinois Inst. of Technol., Chicago, IL 60616 and Dept. of Eng. Sciences, Tech. Univ. of Crete, Chania 73100, Greece)

Elastic interfacial waves propagating along one of the planar boundaries separating an orthotropic interlayer of arbitrary uniform thickness from an orthotropic infinite surrounding solid are studied. The axes of material symmetry of the two materials are aligned with one of the axes coinciding with the propagation direction and another being perpendicular to the interfaces. The dispersion equation is derived in explicit form yielding the interfacial phase or group speed in terms of frequency, nondimensionalized with respect to the interlayer thickness, and the elastic constants and mass densities of the interlayer and the surrounding solid. Limiting cases of the dispersion equation give the secular equation for interfacial (Stoneley) waves in two semi-infinite orthotropic materials and the frequency equation for an orthotropic plate. Analysis of the dispersion equation reveals several features. Under parameter conditions which are defined propagation at low frequencies cannot occur. Also material parameter combinations are found for which interfacial waves of arbitrary wavelength as compared to the interlayer thickness cannot propagate. Finally, the existence of standing waves as solutions of the bifurcation equation, a special case of the dispersion equation, is defined with respect to the parameters of the interlayer and the host material.

10:30

2aPAb8. Scattering of an obliquely incident wave by a transversely isotropic cylinder. Farhang Honarvar and Anthony N. Sinclair (Dept. of Mech. Eng., Univ. of Toronto, 5 King's College Rd., Toronto, ON M5S 3G8, Canada)

While the scattering of a plane acoustic wave from a solid isotropic cylinder has been extensively studied for the past several decades, very little has been investigated regarding scattering from anisotropic cylinders. In this paper, the mathematical formulation for the scattering of a plane acoustic wave incident at an arbitrary angle α on an infinite transversely isotropic cylinder is developed. The normal mode solution is based on decoupling of the SH wave from the P and the SV waves. The resulting partial differential equations have closed-form solutions and the scattered pressure field can be calculated at any point outside the cylinder. The solution degenerates to the well-known simple model for isotropic cylinders in the case of very weak anisotropy. The validity of the mathematical model is verified first by applying it to the familiar case of an isotropic aluminum cylinder. The model is then applied to a transversely isotropic cylinder generated by perturbing various elastic constants of the isotropic aluminum cylinder. The manner in which these perturbations affect each of the Rayleigh, Whispering Gallery, and longitudinally guided wave modes is shown to be consistent with elasticity theory and modal shapes of these resonances. Some experimental results are also presented.

10:45

2aPAb9. The Kirchhoff–Helmholtz integral theorem and related identities for waves in an inhomogeneous moving fluid. Oleg A. Godin (School of Earth and Ocean Sciences, Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 2Y2, Canada)

The Kirchhoff–Helmholtz integral theorem (KHT), which expresses the wave field in a volume inside (or outside) a given surface in terms of the field's value on the surface, is well known and widely used in acoustics of motionless media. In this paper, an extension of the theorem to acoustic-gravity waves in an arbitrary inhomogeneous *moving* fluid is obtained as a corollary of the reciprocity-type relations constituting the recently established flow reversal theorem (FRT) [O. A. Godin, J. Acoust. Soc. Am. **97**, 3396(A) (1995); **98**, 2866(A) (1995)]. The KHT takes a concise form when stated in terms of acoustic pressure and oscillatory displacement of fluid particles. The KHT is applied to study uniqueness of solutions to acoustic boundary value problems in moving media and to establish unitarity and other general properties of the scattering matrix for surface and volume scattering as well as in an irregular (range-dependent) waveguide with flow. Relation of the scattering matrix properties to the FRT and to wave-action conservation is discussed. [Work supported by NSERC.]

11:00

2aPAb10. Sound propagation in a periodically lined duct with a fluid flow. Susann Boij (Dept. of Mech. Eng., Rm. 3-445, MIT, Cambridge, MA 02139) and Börje Nilsson (KTH, 100 44 Stockholm, Sweden)

Resonant lining is an effective measure to reduce tonal noise in ducts. It has been observed that a fluid flow in the duct can interact with the lining in such a way that sound is amplified. A model for the acoustic performance of an infinite duct with periodic lining (quarter wavelength resonators) and a nonzero mean velocity is developed. The fluid in the duct and lining are separated by an infinitely thin shear layer, the velocity in the duct is uniform, and the velocity in the lining is zero. Such a layer is unstable for all frequencies, a so-called Helmholtz' instability. The linear acoustic equations are assumed to be valid. Based on the building block method, the acoustic modes in the periodic structure are calculated from the duct modes and scattering effects of the sharp edges. Numerical computations are done for a number of cases, where higher-order, evanescent, acoustic Floquet modes are omitted in order to simplify calculations. The direction of propagation of the computed modes is determined by the requirement of causality.

11:15

2aPAb11. Giant sonic stop bands in 3-D periodic array of liquid balloons. Manvir S. Kushwaha and Rama N. Singh (Instituto de Fisica, Universidad Autonoma de Puebla, Apdo. Post. J-48, Puebla 72570, Mexico)

Periodic binary systems can create complete acoustic bandgaps (i.e., stop bands) within which sound and vibrations are forbidden. This is of interest for applications such as acoustic filters, noise control, and improvements in transducers; as well as for pure physics concerned with the Anderson localization of sound and vibrations. The band structures for 3-D periodic arrays of water balloons surrounded by mercury have been computed. Complete, multiple stop bands are found for fcc, bcc, and sc lattices. It is stressed that such a simple 3-D inhomogeneous system exhibits the widest stop bands ever reported for elastic as well as for dielectric com-

posites. The gap/mid-gap ratios are found to be as high as ~ 1. For mercury balloons surrounded by water the gaps are found to be comparatively much smaller. Physical importance and applications of such gaps will be discussed.

11:30

2aPAb12. Diffuse field decay curvature for material characterization. John Burkhardt (Dept. of Eng., Purdue Univ., 2101 Coliseum Blvd. E., Ft. Wayne, IN 46805)

A new quantitative technique is proposed for determining both the concentration and spatial extent of viscous dissipation in dynamic systems. When dissipation is distributed unevenly within a system, nonexponential response decays can result within certain frequency ranges. Unlike typical exponential decays that carry volume averaged information about the strength of dissipation, nonexponential decays carry information about both the strength and distribution of dissipation in the system. Two parameters describe the nonexponential decay process and can be directly related to the strength and spatial extent of the dissipation. It is proposed that fitting observed nonexponential decays to the predicted model may provide a method for the nondestructive determination of both the extent and severity of dissipation causing defects in materials. The presented work concentrates on acoustic systems and provides a test of the proposed concept. Theoretical predictions are presented and compared with numerical experiments to determine the effectiveness of the proposed approach.

11:45

2aPAb13. On the reflection of elastic waves in monoclinic incompressible materials. Dimitrios A. Sotiropoulos and Sudhakar Nair (Dept. of Mech., Mater., and Aerospace Eng., Illinois Inst. of Technol., Chicago, IL 60616)

The reflection of plane elastic waves from a free surface of monoclinic incompressible materials is examined under plane strain conditions in a plane of material symmetry. The propagation condition is derived which together with the law of reflection yields an inequality that defines the range of existence of the two (one homogeneous and the other homogeneous or inhomogeneous) reflected waves in terms of the angle of incidence of a homogeneous wave, the orientation of the free-surface with respect to a material axis of symmetry, and the elastic constants of the monoclinic material. The critical orientation beyond which there exist two homogeneous reflected waves is derived in explicit form in terms of the elastic constants. One of these reflected waves has an angle of reflection equal to the angle of incidence only for specific orientations which are found. In the range of existence of the two reflected waves, exclusion points are defined for which there exists only one reflected (homogeneous) wave with nonzero amplitude.

12:00

2aPAb14. Ultrasonic velocity and attenuation of nano-scaled copper measured by laser ultrasonic technique. Xiaorong Zhang, Changming Gan, Shiyi Zhang (State Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Nanjing 210093, China), Yuying Hauny, and Dingchang Xian (Lab. of Synchrotron Radiation, Inst. of High Energy Phys., Chinese Acad. of Science, Beijing 100039, China)

The velocity and attenuation of an ultrasonic longitudinal wave for nano-scaled copper are determined by a laser ultrasonic technique. The nano-scaled copper samples are composed of super fine particles 10-nm in size, and are prepared by a suppressing and sintering technique under a vacuum, and different pressures are used in the experiment. These samples are of thicknesses between 125–300 μm. The experimental results show that the velocity dispersions and attenuation spectra of nano-scaled copper depend on their fabrication technology conditions, which are different from those of conventional Cu. The attenuation of nano-scaled copper is proportional to the frequency of ultrasound, and some absorption peaks appear at the curves of attenuation versus frequency, but the attenuation of conventional Cu is proportional to the square of the frequency. The velocity of nano-scaled copper is lower than that of conventional Cu. The experimental system, measurement method, results, analyses and discussions are also presented.

TUESDAY MORNING, 14 MAY 1996 REGENCY A AND B, 8:30 TO 11:50 A.M.

Session 2aPP

Psychological and Physiological Acoustics: Psychoacoustics and David M. Green

William A. Yost, Cochair
Parmly Hearing Institute, Loyola University, 6525 North Sheridan Avenue, Chicago, Illinois 60626

Ervin R. Hafter, Cochair
Department of Psychology, University of California, 3210 Tolman Hall #1650, Berkeley, California 94720

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. Effects of modulator phase for comodulation masking release and modulation detection interference tasks. Virginia M. Richards and Emily Buss (Dept. of Psych., Univ. of Pennsylvania, 3815 Walnut St., Philadelphia, PA 19104)

Three experiments evaluated the joint effects of target/masker frequency separation and modulator phase on modulation detection interference (MDI) and comodulation masking release (CMR). The tasks were (a) detection of the sinusoidal amplitude modulation (SAM) of a tone, (b) discrimination of changes in modulation depth of a SAM tone, and (c) detection of a tone added to a narrow band of noise. Thresholds were obtained for either the target alone, or the target presented with two maskers. The target frequency was 1500 Hz. The maskers were above and below the target at frequency separations ranging from 1/3 to 2 oct. The maskers were either tones or maskers appropriate for the task (SAM tones or noise). For nontonal maskers, the masker envelopes were either coherent or random with respect to the envelope at the target frequency. Substantial individual differences were obtained. When the tonal maskers led to only modest masking, effects of target/masker envelope randomization tended to occur for frequency separations less than an octave. The results suggest a dependence on interactions within a single, relatively broadband analyzer. [Work supported by NIH and Univ. Penn. Research Foundation.]

9:05

2aPP2. Individual differences in human sound localization behavior. Frederic L. Wightman (Waisman Ctr. and Dept. of Psych., Univ. of Wisconsin, Madison, WI 53705) and Doris J. Kistler (Univ. of Wisconsin, Madison, WI 53705)

Results from studies of the apparent positions of virtual and real sound sources conducted over the past 5 years reveal large individual differences. To characterize these differences in global terms such as accuracy or variance does not tell the whole story. Listeners differ in the extent to which they make front–back or back–front confusions, the extent to which they report apparent positions above or below the horizontal plane, the resistance of their judgments of apparent position to interaural level imbalance, and other factors. It seems clear that some of these differences might be traced to differences in outer-ear acoustics. Since pinna shapes vary tremendously from person to person, there are large individual differences in the resulting head-related transfer functions (HRTFs). The cues provided by the direction-dependency of HRTFs are known to be important, especially for determining apparent source elevation. HRTFs lacking in detail provide ineffective cues and thus listeners must adopt differing strategies for combining HRTF cues with other cues to determine source position. Unfortunately detailed analyses of both HRTF differences and differences in localization behavior does not reveal a simple relationship. [Work supported by NASA and NIDCD.]

9:35

2aPP3. Time for a new form of temporal integration. Roy D. Patterson (MRC Appl. Psych. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK)

Dave Green has repeatedly emphasized the paradox in the leaky integrator model of temporal processing and stressed the need for new approaches. Recent experiments at APU on *envelope* asymmetry, *period* jitter, and *fine-structure* regularity reinforce the paradox. When "damped" noise with a repeating exponential *envelope* is time reversed ("ramped" noise), it alters the envelope and carrier percepts, supporting damped/ramped discrimination for envelope periods from 8–125 ms. The Weber fraction for jitter on the *period* of a click train peaks around 60 ms; perfectly regular click trains sound irregular in this region reducing jitter detection relative to longer and shorter periods. These experiments suggest a long time constant for the leaky integrator. When iterated rippled noise masks random noise, it is 15 dB less effective than the equivalent random noise; low-level random noise disrupts the *fine-structure* regularity of iterated noise. This suggests a short time constant, since values over 8 ms remove the fine-structure that appears to underly the timbre discrimination. "Strobed temporal integration" offers a potential solution to the primary paradox; the fine-structure of periodic sounds is preserved in an "auditory image" which decays relatively slowly thereafter. [Research performed at APU in collaboration with Drs. Akeroyd, Allerhand, Datta, Handel, Irino, Lorenzi, Tsuzaki, and Yost.]

10:05–10:20 Break

10:20

2aPP4. Altered temporal and spectral patterns produced by cochlear implants: Implications for psychophysics and speech recognition. Robert V. Shannon, Fan-Gang Zeng, and John Wygonski (House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

Cochlear implants produce highly unnatural temporal and spatial (tonotopic) patterns of neural activation in the auditory system, and the patterns may vary considerably from patient to patient. In spite of these abnormal patterns, psychophysical measures of temporal processing are relatively normal in patients with cochlear implants and brainstem implants. A high level of speech recognition is possible with only three bands of modulated noise [Shannon *et al.*, Science **270**, 303–304 (1995)]. What parameters of the peripheral activity pattern are critical for the transmission of speech pattern information? Four-band speech processors were constructed which reduced the spectral information in speech to four, amplitude-modulated noise bands. Variations were constructed that altered the location, spectral extent, overlap, and amplitude compression of the four noise bands relative to the original speech. Manipulations of the band overlap and the cut-off frequencies dividing the bands had relatively little effect on speech recognition. However, amplitude compression and the tonotopic misalignment of noise bands with the speech cues resulted in severe reductions in speech recognition. These experiments and the performance levels of present cochlear implant patients demonstrate that normal temporal information and considerable speech information can be received with relatively poor tonotopic selectivity. [Work supported by NIDCD.]

10:50

2aPP5. The index of interaural correlation: Accounting for binaural detection across center frequency. Leslie R. Bernstein (Surgical Research Ctr., Dept. of Surgery, Otolaryngology, and Ctr. for Neurological Sciences, Univ. of Connecticut Health Ctr., Farmington, CT 06030)

Investigations in this laboratory demonstrated that the normalized correlation accounted well for listeners' performance in binaural tasks utilizing high-frequency stimuli for which the envelopes convey the interaural information. This outcome means that the *non-zero* mean of the envelopes must be included in the computation of the correlation. Due to peripheral auditory processing analogous to rectification and low-pass filtering, the "internal" representations of auditory stimuli have nonzero means regardless of their frequency. This study evaluated whether the normalized correlation computed subsequent to rectification and low-pass filtering could account for detection data at low and intermediate, as well as high frequencies. In a four-interval, two-alternative task, listeners detected which interval contained a tone (between 500 Hz and 2 kHz) added antiphasically to diotic, 100-Hz-wide, noise (NoSπ). "Nonsignal" intervals contained the tone added homophasically (NoSo). Performance was measured for S/Ns between -30 and $+30$ dB. For all

S/Ns, overall level was 70 dB SPL. Normalized correlation (unlike the correlation *coefficient*) described performance well, as a function of S/N and center frequency. The parameters of the rectifier and low-pass filter required to describe the data will be discussed. [Supported by research Grant No. 5R01DC0210302, National Institute on Deafness and Other Communication Disorders, NIH.]

11:20

2aPP6. Temporal cues in monaural and binaural detection and discrimination. David A. Eddins (Psychoacoust. Lab., Dept. of Speech and Hearing Sciences, Indiana Univ., Bloomington, IN 47405)

The auditory system can extract temporal information from complex stimuli to perform a variety of detection and discrimination tasks. At low stimulus frequencies, the auditory system may encode stimulus fine structure and temporal envelope. At higher frequencies, the coding of fine structure diminishes as phase locking deteriorates, while the coding of the temporal envelope remains. The present study investigated the relative contributions of interaural differences in stimulus envelope and fine structure to the masking level difference (MLD). NoSo and NoS thresholds were measured at stimulus frequencies of 500 and 4000 Hz using two types of narrow-band (50 Hz) noise: Gaussian and low-noise noise. Gaussian noise maskers yielded MLDs of 15 at 500 Hz and 9 dB at 4000 Hz, the reduction reflecting minimal fine structure cues at 4000 Hz. Low-noise noise yielded lower NoSo and higher NoS thresholds than Gaussian noise. MLDs of 8 at 500 Hz and 0 dB at 4000 Hz reflected minimal masker envelope cues in both conditions and reduced fine structure cues at 4000 Hz. While the pattern of results was consistent, threshold differences among the six listeners were substantial. Additional MLD conditions and data from CMR and noise discrimination tasks employing low-noise noise will be discussed.

TUESDAY MORNING, 14 MAY 1996 NATIONAL PARKS, 8:30 TO 11:30 A.M.

Session 2aSC

Speech Communication: Models and Measurements

Marios S. Fourakis, Chair

Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey, Columbus, Ohio 43210-1372

Contributed Papers

8:30

2aSC1. Iceberg revisited. Osamu Fujimura (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210-1002)

Iceberg patterns refer to a particular part of the flesh-point coordinate time function of a crucial articulator for an obstruent gesture [O. Fujimura and W. Spencer, J. Acoust. Soc. Am. Suppl. 1 **76**, S59 (1983)]. They were found stable for a given demisyllable in different prosodic contexts, when a microbeam pellet height value, around which the time function was excised for comparison, was optimally selected. The time for crossing such a selected threshold height was found useful for studying phonetic phrasing patterns. The C/D model [O. Fujimura, J. Acoust. Soc. Jpn. (E) **13**, 39–48 (1992); O. Fujimura, Proc. ICPhS **3**, 10–17 (1995)] proposes that the articulatory control function is composed of impulse response functions (IRFs) for elemental gestures, which are magnified according to the syllable strength. Therefore, the movement velocity at a given time relative to the syllable pulse (i.e., in a given phase of transition) will vary according to the syllable magnitude. The two approximation principles will be shown to be mutually consistent when the saturation effects of the physical articulatory system is considered along with general properties of the IRFs for obstruent gestures. [Work supported in part by research and education funds provided by ATR/ITL and ATR/HIP.]

8:45

2aSC2. Across session variability in lip–jaw synergies for bilabial closure. Peter J. Alfonso (Dept. of Speech and Hearing Sci., Univ. of Illinois, 901 S. Sixth St., Champaign, IL 61820)

Movements of the tongue, lips, and jaw were transduced by electromagnetic midsagittal articulography. A single session included 20 repetitions of \pap\,\tat\, and \sas\ imbedded in a carrier phase at normal, slow, and fast speech rates. Seven talkers completed three sessions at 1-week intervals. Reported here are across-session comparisons of lip–jaw displacements, velocities, temporal ordering, and relative timing for bilabial closure at normal rates. Four of the seven subjects showed stable but idiosyncratic spatial and temporal organizational characteristics within and across sessions. Subjects who are relatively unstable across sessions in regard to displacement characteristics are also relatively unstable in regard to temporal ordering and relative time. Only motor equivalence covariability is stable across sessions for all subjects, that is, even in the case of unstable spatial and temporal organizational characteristics. Results seem to support a task-dynamic point of view, namely, that the spatial and temporal organizational characteristics of the individual articulators that comprise an articulatory complex represent the natural consequence of gestural coordination (inferred here by motor equivalence index) and therefore would not demonstrate stability across sections. [Work supported by NIH DC-00121 to Haskins Laboratories and University of Illinois research grant.]

9:00

2aSC3. Glottal geometry and phonation threshold pressure in a vocal fold physical model. Roger W. Chan, Ingo R. Titze, and Michael R. Titze (Dept. of Speech Pathol. and Audiol. and Natl. Ctr. for Voice and Speech, Univ. of Iowa, Iowa City, IA 52242)

The effect of convergent and divergent prephonatory glottal geometries on phonation threshold pressure was investigated using a previously reported physical model of the vocal fold mucosa [Titze *et al.*, J. Acoust. Soc. Am. **97**, 3080–3084 (1995)]. Lowest offset phonation threshold pressure in the range of 0.07 to 0.23 kPa was obtained for a rectangular or a near-rectangular glottis (with convergent or divergent angle $\leqslant 3°$) across different glottal half-widths between 2.0 and 6.0 mm. Phonation threshold pressure for more convergent or divergent glottal geometries was consistently higher. This finding only partially agrees with previous analytical

work which predicts a lowest phonation threshold pressure for a divergent glottis [I. R. Titze, J. Acoust. Soc. Am. **83**, 1536–1552 (1988) and J. C. Lucero, J. Acoust. Soc. Am. **98**, 779–784 (1995)]. The discrepancy between theory and data is likely to be associated with flow separation from a divergent glottis. [Work supported by NIH.]

9:15

2aSC4. Effects of driving pressure and recurrent laryngeal nerve stimulation on glottic vibration in an *in vivo* constant pressure model of phonation. Andrew Verneuil, Jody Kreiman, Kevin Kevorkian, Ming Ye, Bruce R. Gerratt, and Gerald S. Berke (Div. of Head and Neck Surgery, UCLA School of Medicine, CHS 62-132, Los Angeles, CA 90095)

Most *in vivo* studies of phonation have used constant flow models. However, because the lung–thorax system is better viewed as a constant pressure source, a constant pressure *in vivo* canine model was developed. In this model, driving pressure and recurrent laryngeal nerve (RLN) stimulation are independent variables, and subglottic pressure (P_{sub}; measured immediately under the glottis), air flow, fundamental frequency ($F0$), and glottic area are dependent variables. In three dogs, P_{sub} and air flow were measured with constant RLN stimulation and varying driving pressure, and in another condition with constant driving pressure and varying RLN stimulation. Videostroboscopic measures on four animals assessed glottic areas with constant RLN stimulation and varying driving pressure. With constant RLN stimulation, increasing driving pressure had no effect on glottic area, while $F0$, and air flow increased significantly. Changes in P_{sub} were small in comparison to changes in driving pressure. With constant driving pressure, increasing RLN stimulation increased P_{sub} and $F0$ and decreased air flow. These findings suggest that P_{sub} during phonation is primarily dependent on RLN activity and associated laryngeal muscular contraction, but not on lung driving pressure. [Work supported by VA Merit Review funds.]

9:30

2aSC5. Annealing in a genetic algorithm for task-dynamic recovery from speech acoustics. Richard S. McGowan (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Genetic algorithms, as they have been used for recovery of task-dynamic parameters [R. S. McGowan, "Recovering articulatory movement from formant frequency trajectories using task dynamics and a genetic algorithm," Speech Commun. **14**, 19–49 (1994)], can take some time to converge, which means that expensive function evaluations are necessary to obtain optimum solutions. In previous experiments, each task dynamic- target was always given 6 bits in the chromosome. In the tests reported here, the number of bits in the chromosome string representing target positions was made a function of generation, with the range of the targets remaining constant to see whether an annealing process with ever increasing resolution would speed convergence to a good fitness maximum. Results show that annealing can be helpful in speeding convergence, although not all annealing schedules help. This procedure is related to simulated annealing where the temperature of the search is decreased as the search progresses. It is also related to codebook look-up used in the inverse problem, where vocal tract shapes that produce speech close to that of the data are accessed to initialize a subsequent optimization procedure. [Work supported by NIH Grant DC-01247 to Haskins Laboratories.]

9:45–10:00 Break

10:00

2aSC6. Labial kinematics in stop consonant production. Anders Lofqvist and Vincent L. Gracco (Haskins Labs., 270 Crown St., New Haven, CT 06511)

This study examines kinematic patterns of upper and lower lip movements in bilabial stop production with particular emphasis on events before and during the oral closure. Lip movements were recorded using a magnetometer system. Five subjects participated and produced 50 tokens of the sequences /aCV/, where C was either /p/ or /b/ and V one of the vowels /i, a, u/. Force of labial contact and oral air pressure were also recorded in one subject. The results indicate that at the instant of oral closure, the velocity of both lips was close to peak velocity. They also show that the lower lip continued its upward movement after oral closure. This was reflected in the force of labial contact, which showed an increase in contact pressure after the oral closure had occurred. During the period of lower lip upward movement after the oral closure, the upper lip often showed a corresponding upward movement, suggesting that the lower lip might be pushing the upper lip. These results support the hypothesis that one goal in making the lip closure is a region of negative lip aperture. [Work supported by NIH.]

10:15

2aSC7. Lip protrusion movements in normal versus clear speech. Pascal Perrier, Joseph S. Perkell, and Melanie L. Matthies (Res. Lab. of Electron., Rm. 36-511, MIT, Cambridge, MA 02139)

A kinematic analysis of midsagittal movement (EMMA) data was performed for upper lip protrusion from [i] to [u] in [iku], [ikku], and [iktku], produced by a speaker of American English. Utterances were embedded in a carrier sentence and spoken in four conditions: normal rate, normal rate and clear, fast rate, fast rate and clear. The results show several regularities across all conditions: (1) the delay between the protrusion onset and the acoustic onset of [u] varies linearly with the vowel-to-vowel duration (VVD), [i]-end to [u]-onset, with a slope of 0.7; (2) the overlap between the protrusion movement and vowel [i] decreases as VVD increases, with no significant differences between conditions for similar VVD values; (3) the relationship between the ratio of maximum velocity (V_{max}) to protrusion amplitude (Amp) and the movement duration (T) is constant: $V_{max}/\mathrm{Amp}=c/T$; (4) the number of velocity peaks increases linearly with VVD. These regularities suggest that the same control strategy underlies the protrusion gesture, whatever the condition. However, for a given rate, clear tokens were produced with longer VVDs and larger protrusion amplitudes. For similar VVD values, fast clear tokens had larger values of V_{max} than normal tokens. [Work supported by NIDCD and NATO.]

10:30

2aSC8. Variation of subglottal pressure during sentence production: Effect on glottal source amplitude and spectrum. Helen M. Hanson (Sensimetrics Corp., 26 Landsdowne St. and Res. Lab. of Electron., MIT, 50 Vassar St., Cambridge, MA 02139)

With the goal of synthesizing natural-sounding speech based on higher-level parameters, variations in subglottal pressure for sentences having different stress patterns have been studied. A Rothenberg mask was used to collect oral pressure and airflow signals from several subjects producing reiterant speech. Acoustic sound pressure was also recorded, and sound pressure levels (SPLs) for both reiterant and natural utterances were compared to verify the suitability of reiterant speech for the task. Subglottal pressure P_s was inferred from oral pressure measured during stop consonant closure of reiterant speech, average glottal flow U_g was estimated from the low-pass-filtered airflow signals, and average glottal area A_g was calculated using P_s and U_g. Glottal source amplitude and spectral tilt were estimated from the speech waveform. Preliminary analysis shows that accented syllables are marked by an increase in P_s of about 50% relative to the weakest unaccented syllables, an increase in SPL of 10 to 15 dB, and some increase in A_g. A model for estimating glottal parameters during these utterances has been developed. Attempts to synthesize sentences based on this model, by including A_g and P_s as control parameters, are described. [Work supported by NIH grants DC00075 and MH52358.]

10:45

2aSC9. A nonlinear finite-element model of the vocal fold. Arthur P. Lobo and Michael O'Malley (Berkeley Speech Technologies, 2246 Sixth St., Berkeley, CA 94710)

A large-displacement large-strain 3-D finite-element model of the vocal fold was developed. The structure is discretized into 720 elements with 3003 displacement and 720 pressure degrees of freedom. The model incorporates material and geometric nonlinearities. For the constitutive law, the Mooney–Rivlin rubber material formulation for an anisotropic hyperelastic material is used. Average incompressibility constraints are introduced by adding a hydrostatic pressure work term (Lagrange multiplier) to the strain energy density function. This term is a function of the bulk modulus which has the numerical equivalence of the penalty parameter. The nodal displacements and pressure are solved for independently, using a mixed displacement/pressure formulation with 8 displacement nodes (trilinear/hexahedron) and a constant (uniform) pressure term per element. Static condensation of the discontinuous pressure variable at the element level keeps the half-bandwidth of the stiffness matrix the same as for the displacement-only formulation. All nodes on the anterior commissure, vocal processes and the lateral surface (attached to the thyroid cartilage) are fixed-essential boundary conditions. An incremental-iterative strategy solves the dynamic equilibrium equations of motion in the total Lagrangian formulation. The vocal fold deformation was studied under a periodically time-varying pressure profile (natural boundary conditions) applied on 117 medial surface nodes.

11:00

2aSC10. Tongue control in speech: Some proposals tested on a biomechanical model. Yohan Payan and Pascal Perrier (Institut de la Communication Parlee, INPG & Univ. Stendhal, 46 Av. Felix Viallet, F38031 Grenoble, France)

The ability of a 2-D biomechanical model of the tongue to reproduce kinematic properties of human tongue movements in V-V transitions is assessed. The model consists of a finite elements description, and simulates the main agonist/antagonist muscle pairs (posterior Genioglossus/Hyoglossus and anterior Genioglossus/Styloglossus). The external geometrical shape of the model has been adapted from X-ray data, to fit the tongue contours of a French native speaker. Movement data were collected on this speaker with an electromagnetometer (Carstens Electronics, AG100) for the tokens [i-a], [i-e], [i-ε], [a-i], [ε-i], [e-i], [i-e-ε-a], [a-ε-e-i], [y-œ], [y-o], [y-u], [u-y], [o-y], [œ-y], [y-œ-o-u], [u-o-œ-y]. For each token, the speaker was asked to keep the lip shape constant. The jaw was held in a constant position with a small bite-block (5 mm). For the simulations, according to Feldman's equilibrium-point hypothesis [A. Feldman, "Once more Equilibrium-Point Hypothesis for motor control," J. Motor Behav. **18** (1986)], movement was produced by shifting, at a constant rate, the mechanical equilibrium of the motor system from one equilibrium-target position to another one. Simulations are assessed in the kinematic and acoustic spaces. [Work supported by the European Union.]

11:15

2aSC11. The effect of lexical complexity on segmental intelligibility. Howard C. Nusbaum and Alexander L. Francis (Dept. of Psych., Univ. of Chicago, 5848 S. Univ. Ave., Chicago, IL 60637)

Most intelligibility tests are based on the use of monosyllabic test stimuli. This constraint eliminates the ability to measure the effects of lexical stress patterns, complex phonotactic organizations, and morphological complexity on segmental intelligibility. Since these aspects of lexical structure affect speech production (e.g., changing syllable duration), it is likely that they affect the structure of acoustic-phonetic patterns. It also seems plausible that listeners make use of this knowledge during segmental perception. Thus, to the extent that text-to-speech systems fail to modify acoustic-phonetic patterns appropriately in polysyllabic words, intelligibility may suffer. This means that while most standard intelligibility tests may accurately estimate segmental intelligibility in monosyllabic words, this estimate may not generalize prediction of segmental intelligibility with more complex lexical forms. The present study was carried out to measure segmental intelligibility in stimuli varying in lexical complexity. Monosyllabic, bisyllabic, and polysyllabic words were used varying in morphological complexity (monomorphemic or polymorphemic). Listeners transcribed these stimuli spoken by two human talkers and two text-to-speech systems varying in speech quality. The results indicate that lexical complexity does affect the measured segmental intelligibility of synthetic speech.

TUESDAY MORNING, 14 MAY 1996 REGENCY D, 8:00 TO 11:50 A.M.

Session 2aUW

Underwater Acoustics: Low Grazing Angle Acoustic Penetration of the Bottom at a Kilohertz and Above

Eric I. Thorsos, Chair

Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Chair's Introduction—8:00

Invited Papers

8:05

2aUW1. Observations of anomalous acoustic penetration into sediment at shallow grazing angles. Joseph L. Lopes (Coastal Systems Station Dahlgren Div., Naval Surface Warfare Ctr., Code 130B, 6703 West Hwy. 98, Panama City, FL 32407-7001)

Buried mines pose a difficult problem for mine countermeasures operations. Mines may naturally bury in coastal waters where wave-induced effects are significant near the bottom. Current techniques for acoustic detection and classification of a buried mine require a high-resolution imaging sonar with bottom penetrating capability. Recently, anomalous acoustic penetration into sediment at shallow grazing angles was observed during a series of experiments which acoustically characterized the shallow water environment. These experiments employed a stationary sonar tower complete with pan and tilt motors. In the initial experiments, backscattered returns from objects of opportunity were recorded; these objects were either partially or completely buried near the water–sediment interface. In a later experiment, buried calibrated retroreflectors and a buried hydrophone were also utilized; the hydrophone was used to measure in-sediment sound-pressure levels. Results of the backscattered returns from the objects of opportunity and calibrated retroreflectors will be presented. Also, the measured in-sediment sound-pressure levels will be compared to the SAFARI (seismic and acoustic fast field algorithm for range-independent environments) code and will be further compared to results reported in the literature. [Work supported by the Office of Naval Research.]

8:30

2aUW2. Acoustic penetration of ocean sediments in the context of Biot's theory. Nicholas P. Chotiros (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Acoustic penetration of ocean sediments will be examined in the light of the Biot–Stoll theory of sound propagation in porous media. This theory is attractive from a number of perspectives. From an applications point of view, it can account for a number of anomalies that have been observed experimentally but are not explainable in terms of the simpler viscoelastic wave propagation theory. From a theoretical perspective, it provides improved connections between the sediment geophysical properties, such as grain size, porosity, permeability, and frame moduli, and the acoustic properties, such as reflection loss, sound speeds, and attenuations. The difficulty in using Biot's theory is mostly in the understanding and quantification of certain key input parameters, particularly the elastic moduli of the solid matrix, and of the constituent granular material. The potential reward is a unified theoretical frame work for research into ocean sediment acoustics that brings acoustics and geophysics closer together. [Work supported by ONR, Code 321 OA.]

8:55

2aUW3. Subcritical penetration of sandy sediments due to interface roughness. Darrell R. Jackson, John E. Moe, Eric I. Thorsos, and Kevin L. Williams (Appl. Phys. Lab., College of Ocean and Fishery Sciences, Univ. of Washington, Seattle, WA 98105)

Seafloor roughness can cause acoustic energy to propagate into sediments for "subcritical" incident grazing angles, that is, grazing angles smaller than the critical angle determined by the compressional wave speed. Such effects must be considered in interpreting experimental data on sound penetration into sediments. The sediment is modeled as a fluid supporting only compressional waves and a twofold theoretical approach is used. First, an exact integral equation is solved numerically to obtain the penetrating field in two dimensions. Physical insights are abstracted from these results which are also used to show that perturbation theory is valid for problems of interest. Using perturbation theory, a numerically tractable three-dimensional model is compared to some of the experimental data of Chotiros and colleagues [N. P. Chotiros, J. Acoust. Soc. Am. **97**, 199–214 (1995)]. The model and data match well subject to plausible assumptions about roughness statistics. [Work supported by ONR.]

Contributed Papers

9:20

2aUW4. Poroacoustic media. Michael D. Collins (Naval Res. Lab., Washington, DC 20375) and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, New York 12180)

Some poroelastic sediments have relatively high slow-wave speeds and low shear-wave speeds [N. P. Chotiros, J. Acoust. Soc. Am. **97**, 199–214 (1995)]. Wave propagation in these kinds of sediments may be modeled efficiently by neglecting shear waves to obtain the equations of motion of poroacoustic media. With this approach, the number of equations is reduced and numerical implementations may use larger grid spaces. Poroacoustic waves satisfy two coupled equations. Poroelastic waves satisfy three coupled equations for two-dimensional problems (the original formulation involves four equations). The equations of poroacoustics are symbolically identical to the equations of acoustics. The poroacoustic wave equation is a vector generalization of the variable density acoustic wave equation [P. G. Bergmann, J. Acoust. Soc. Am. **17**, 329–333 (1946)]. The interface conditions between poroacoustic layers are vector generalizations of the interface conditions between fluid layers. The energy flux integral of poroacoustics is a vector generalization of the energy flux integral of acoustics. The equations of motion are in a form suitable for factoring and solving with parabolic equation techniques. [Work supported by ONR.]

9:35

2aUW5. Energy-conservation conditions for poroelastic and poroacoustic waveguides. Joseph F. Lingevitch, Michael D. Collins (Naval Res. Lab., Washington, DC 20375), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Wave propagation in range-dependent ocean environments is often treated by dividing the domain into a sequence of range-independent regions. One-way techniques, such as the parabolic equation method, are applied to efficiently solve the wave equation in each region. The reduced order of one-way wave equations limits the number of boundary conditions that may be imposed at the vertical interfaces between regions. This can result in significant errors as well as instabilities. Accurate solutions may be obtained for many problems by a conserving energy flux at the vertical interfaces [M. B. Porter *et al.*, J. Acoust. Soc. Am. **89**, 1058–1067 (1991)]. The standard expressions for energy flux are of limited practical use because they are nonlinear. It is possible to obtain equivalent linear conditions by writing the energy flux integrand as a perfect square involving roots of operators. The acoustic case has been resolved and some progress has been made for the elastic case. Problems currently under investigation include energy-flux conditions for poroelastic and poroacoustic media as well as corrections for problems involving different types of layers. [Work supported by the NRC and ONR.]

9:50

2aUW6. A modification of Biot theory for unconsolidated sediment. K. Miguel Nathwani and Frank S. Henyey (Appl. Phys. Lab. and Dept. of Phys., Univ. of Washington, Seattle, WA 98105)

A mechanism for bottom penetration at low grazing angle is conversion of sound into Biot slow waves. Experiments in unconsolidated sediments have not found Biot slow waves with expected parameter values, but have found them in consolidated sediments. These results led us to consider a modification of Biot's model for unconsolidated, sandy sediment. In our model, unstrained grains touch at a single point, since two adjacent grain surfaces have relative curvature. As the grains are pressed together, the contact area and the stress/strain ratio increase. Under extension the grains separate and there is no stress. The model predicts (lattice strain) $\propto(\text{stress})^{1/2}$ under compression. This scaling law should apply to any such medium, but the numerical coefficient depends on detailed properties of the lattice and grain geometry. Slow waves resulting from sound incident on the sediment are studied within the framework of this model. For simplicity, normal incidence is assumed. A positive effective lattice bulk modulus results from sediment overburden or from the pressure field of the sound. Numerical solutions and bounds on the production of slow waves in such a medium imply that the intensity of the slow wave is too low to have been observed by the experiments.

10:05–10:20 Break

10:20

2aUW7. Modeling the effects of shear conversion on low grazing angle bottom penetration. Stanley A. Chin-Bing (Naval Res. Lab., Stennis Space Center, MS 39529-5004) and Joseph E. Murphy (Univ. of New Orleans, New Orleans, LA 70148)

Modeling shear conversion with finite-element/finite-difference codes at a kilohertz and above is difficult due to the number of computational elements needed for such small wavelengths. Typically six to ten nodes per wavelength are required to attain good accuracy. Thus, a typical shallow-water two-dimensional scenario can require hundreds of thousands of computational elements. In this model study, a brute-force hybrid approach is applied. The computational resources of supercomputers with large memories are combined with hybrid modeling techniques to produce simulations of compressional/shear-wave conversions at low grazing angles over range-dependent bathymetry. Examples will be presented that illustrate the effects of these conversions in both horizontally stratified and inhomogeneous anelastic ocean bottoms. [Work supported by ONR/NRL and by High Performance Computing grants.]

10:35

2aUW8. The accuracy of perturbation theory for acoustic penetration of sediment due to interface roughness. Eric I. Thorsos (Appl. Phys. Lab., College of Ocean and Fishery Sciences, Univ. of Washington, Seattle, WA 98105)

Lowest-order perturbation theory (PT) is usually considered valid for modeling scattering from rough surfaces with a small roughness. In particular, for PT to be accurate near the specular direction, it is required that $kh \ll 1$, where k is the acoustic wave number and h is the rms surface height. In the case of an acoustic wave incident on the seafloor, a rough interface will scatter sound back into the water and also into the sediment; such scattering into the sediment will occur even when the incident grazing angle is below the critical angle. Here, the accuracy of PT for acoustic penetration through a rough water/sediment interface with a power-law roughness spectrum is examined using exact integral equation results for the 2D problem. It is found that PT remains accurate for kh up to and well beyond unity, when the incident grazing angle is either above or below the critical angle. The reason for this surprising result will be discussed. [Work supported by ONR.]

10:50

2aUW9. Low grazing angle scattering by a rough ocean floor. Paul E. Barbone (Dept. of Aerospace & Mech. Eng., Boston Univ., Boston, MA 02215) and Mark Spivack (Univ. of Cambridge, Cambridge CB3 9EW, UK)

Low grazing angle acoustic scattering by an irregular ocean floor is considered. The ocean bottom is modeled as a semi-infinite acoustic medium with properties that differ from those of the fluid above. At zero grazing incidence, a flat ocean floor appears acoustically soft. Surface roughness introduces perturbations to both the surface height and the apparent impedance of the interface. This nonuniform limit is examined as a perturbation from a slightly rough pressure release surface. A modified parabolic equation is derived which describes the acoustic amplitude to large distances, and takes into account multiple scattering and radiation into the ocean and bottom. An equivalent formulation will be presented in terms of an integral equation which is solved numerically. Average results as well as results from single realizations are presented. [Work supported by NATO.]

11:05

2aUW10. Bottom penetration at subcritical grazing angles by scattering. Ralph A. Stephen (Woods Hole Oceanograph. Inst., 360 Woods Hole Rd., Woods Hole, MA 02543)

The physics of sound propagation and scattering is distinctly different between hard and soft ocean bottoms. A hard bottom is defined as having a shear velocity greater than water velocity; a soft bottom has a shear velocity less than water velocity. Geologically hard seafloors consist of igneous rocks such as gabbros and basalts; soft seafloors consist of sediments such as mud and sand. For a soft bottom with sound incident from above, total internal reflection never occurs. For all grazing angles there is always a Snell's law ray path from compressional energy in the water to shear energy in the bottom. For hard, flat bottoms over homogeneous rock, total internal reflection will occur for grazing angles less than critical. However in this case sound can penetrate the bottom by scattering from surface roughness or volume heterogeneities. Snapshots from time domain modeling by the finite-difference method show that the scattered field consists of wavefronts which are analogous to transmitted compressional and shear body waves. Since the seafloor is rough and laterally heterogeneous over a broad range of length scales, this mechanism could be significant over a broad frequency range. [Work supported by ONR.]

11:20

2aUW11. Near-field scattering through and from a 2-D fluid-fluid rough interface. John E. Moe and Darrell R. Jackson (Appl. Phys. Lab., College of Ocean and Fishery Sciences, Dept. of Elec. Eng., Univ. of Washington, Seattle, WA 98105)

A general analytical expression for the time-dependent field intensity scattered from or through (penetrating) a 2-D fluid-fluid rough surface due to a narrow-band incident plane wave and a narrow-band point source are derived and expressed in terms of the bistatic scattering cross section per unit area of the rough surface. Even though the scattering cross section is defined as a far-field quantity, this near-field result is general and exact for the special case of a continuous wave source and incident plane wave. Dispersion of a pulse is a function of medium parameters, the incident and scattered directions, as well the bandwidth and center frequency of the source signal. First-order perturbation calculations for the case of a Gaussian pulse illustrate intensity pulse dispersion effects due to forward scattering into a lossy sediment. [Work supported by ONR.]

11:35

2aUW12. Peculiarities of low grazing angle acoustic penetration of the layered absorbing bottom. Margarita S. Fokina (Inst. of Appl. Phys., Russian Acad. of Sciences, 46, Ulyanov Str., N. Novgorod, 603600, Russia)

Acoustic reflection from the bottom and sound propagation are closely connected. In this paper reflection coefficient behavior at high frequencies (above 1 kHz) for the layered absorbing bottom was investigated. Influence of the layered bottom parameters on acoustic propagation was considered. It was shown that for the same set of bottom parameters the sound reflection coefficient at low grazing angles decreases when the grazing angle decreases. Behavior of frequency and angle resonances of the reflection coefficients was analyzed. Two types of bottom connections between sets of bottom parameters and sound losses in a wide frequency band were ascertained.

Session 2pAAa

Architectural Acoustics: The Technical Committee on Architectural Acoustics Vern Knudsen Distinguished Lecture

William J. Cavanaugh, Chair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—3:45

Invited Paper

3:50

2pAAa1. Variance and invariants in room acoustics: A random walk through reverberant fields. David Lubman (David Lubman & Assoc., 14301 Middletown Ln., Westminster, CA 92683)

The understanding of room reverberation has advanced notably in recent decades owing to the merging of statistical communication theory with room acoustics. New invariants expressed as simple and robust statistical laws govern steady-state variation of reverberant sound pressure over time, frequency, and space. Much of statistical room acoustics (SRA) theory can be developed by applying random walk theory to classical room acoustics. Reverberation time (T_{60}) controls the variance of reverberant pressure with frequency at each room location. The "reverberation time-bandwidth" (BT_{60}) product controls spatial variance. SRA contradicts the conventional wisdom that "hot spots" and "dead spots" indicate poor diffusion, implying instead that spatial variations are inevitable even in a perfectly diffuse field. SRA facilitated new measurement standards for estimating pure tone sound power in reverberation rooms by providing objective guidance for optimization of spatial averaging and estimation of error bounds. SRA points to fundamental uncertainties in low-frequency room behavior and connects room acoustics to the kinetic theory of gases. Using statistical theory to solve deterministic problems of room reverberation may seem paradoxical if not heretical. Yet SRA continues to provide unexpected insights and discoveries impacting room acoustic thinking and practice. This paper recounts developments from a research participant's perspective.

TUESDAY AFTERNOON, 14 MAY 1996 MT. RUSHMORE, 4:45 TO 5:45 P.M.

Session 2pAAb

Architectural Acoustics: An Afternoon with Leo Beranek

William J. Cavanaugh, Chair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Sponsored by the Technical Committee on Architectural Acoustics and The National Council of Acoustical Consultants

In 1962 Leo Beranek published his classic study of some 55 concert halls throughout the world... "Music, Acoustics and Architecture." This comprehensive compilation of physical and acoustical data on these halls as well as scaled drawings and photographs so that direct visual comparisons of one hall with another were possible, proved invaluable to architects, acousticians, musicians, symphony orchestras and others contemplating building new or renovating existing halls. Several years ago the Technical Committee on Architectural Acoustics at the urging of Christopher Jaffe voted to invite Leo Beranek to update and reprint the book through the Society's "Books" program. Leo Beranek, in his usual scholarly fashion, decided to do more. He has added many new halls, omitted others, and, most significantly, added a totally new section reviewing the enormous amount of work that has been ongoing these past three decades. The Technical Committee on Architectural Acoustics along with the National Council of Acoustical Consultants is sponsoring this reception on the occasion of the publication of this new work. Leo Beranek will say a few words about his recent studies reported in the book and respond to questions from the audience. But mostly this will be a "first for ASA" opportunity to thank the author for his dedication and contribution to our expanding knowledge base on halls for music performance as well as to just plain socialize and talk about concert hall acoustics.

Session 2pBV

Bioresponse to Vibration and to Ultrasound: General Topics

Janet M. Weisenberger, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, Columbus, Ohio 43210

Chair's Introduction—1:25

Contributed Papers

1:30

2pBV1. Exposure guidelines for Navy divers exposed to low-frequency active sonar. F. Michael Pestorius (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029) and Michael D. Curley (Naval Submarine Medical Res. Lab., Groton, CT 06349-5900)

A 30-month experimental and modeling study was conducted into the effects of low-frequency active (LFA) sonars on Navy divers. 453 dives using 87 subjects were completed with one symptomatic event. A variety of dive equipment, depths to five atmospheres, and three signal waveforms were used. Interim guidance for exposure to LFA signals was issued by the U.S. Navy Bureau of Medicine and Surgery. For exposures to nonimpulsive waterborne sound in the frequency range 160–320 Hz, the following set of parameters did not compromise the safety of a highly trained, fit, and informed diver (diver compromise refers to a diver who could become a casualty or burden to others in an operational setting): A maximum overall SPL of 160 dB for no more than 100 s continuous exposure on a 50% duty cycle, with a cumulative exposure limit of 15 min/dive-day, a total of 9 days of exposure in a 2-week period. SPL is measured at the diver's location. This guidance is generalizable only to those populations equivalent in medical health and fitness to U.S. Navy divers. [Work supported by U.S. Navy.]

1:45

2pBV2. A new multifinger tactual display. Hong Z. Tan and William M. Rabinowitz (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

A multifinger positional display, called the Tactuator, was developed to study communication through the kinesthetic and vibrotactile aspects of the tactual sensory system of the hand. The display consists of three independent single-point actuators interfaced (individually) with the fingerpads of the thumb, the index finger, and the middle finger. Each actuator utilizes a disk-drive head-positioning motor augmented with angular position feedback from a precision rotary variable differential transformer (RVDT). A floating-point DSP system provides real-time positional control, using position and derived-velocity feedback. Stimuli from threshold to about 50 dB SL can be delivered throughout the frequency range from near dc to above 300 Hz, thereby encompassing the perceptual range from gross motion to vibration. Actuator frequency and step responses are well modeled as a second-order linear system. Distortion is low allowing delivery of arbitrary stimulus waveforms, e.g., 25-mm low-frequency motion with superimposed high-frequency vibration. System noise and interchannel cross talk are also small. As one example of behavioral performance verification, absolute thresholds measured with the stimulator are in general agreement with reference values. Overall, the Tactuator accurately follows its drive waveforms and is well suited for a variety of multifinger tactual perceptual studies. [Work supported by NIDCD.]

2:00

2pBV3. Information transmission with a multifinger tactual stimulator. Hong Z. Tan, Nathaniel I. Durlach, William M. Rabinowitz, and Charlotte M. Reed (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

The information transmission capabilities of a multifinger positional stimulator, called the Tactuator, were explored in a series of absolute identification experiments. Variations in the frequency of single sinewave stimuli evoked three relatively distinct perceptual attributes: smooth motion (up to about 6 Hz), a rough or fluttering sensation (about 10 to 70 Hz), and smooth vibration (above about 150 Hz). Multicomponent stimuli were formed by summing sinusoids from each of these three regions, with the intent that frequency (and amplitude) variations within each region could be identified independently. Stimulation was applied to either one of three digits (thumb, index, or middle) or to all three digits simultaneously. For stimulus durations of 500 and 250 ms, information transfer (IT) was 6.6 bits (corresponding to perfect identification of 97 stimuli); at 125 ms, IT was 6.0 bits. Estimates of potential IT rates were obtained by sequencing three random stimuli and (a) having the subject identify only the middle stimulus and (b) extrapolating this IT to that for continuous streams. Estimated IT rates were 13 bits/s, and the optimal stimulus presentation rate was approximately 3 items/s regardless of stimulus duration. This IT rate is roughly the same as that achieved by Tadoma users in tactual speech communication. [Work supported by NIDCD.]

2:15

2pBV4. Active noise reduction stethoscopy for lung sounds measurement in loud environments. Samir P. Patel, George R. Wodicka (School of Elec. and Comput. Eng., Purdue Univ., West Lafayette, IN 47907-1285), and Matthew G. Callahan (Univ. Res. Engineers & Assoc., Inc., Acton, MA)

Auscultation of lung sounds in vehicles such as an ambulance or aircraft is unachievable because of high noise levels. Also, the bandwidths of lung sounds and vehicle noise typically have significant overlap, limiting the utility of bandpass filtering. In this study, a passively shielded stethoscope coupler with one microphone to measure the (noise-corrupted) lung sounds and another to measure the ambient noise was constructed. Lung sound measurements were performed on a healthy subject in a simulated USAF C-130 aircraft environment within an acoustic chamber at noise levels ranging from 80 to 100 dB SPL. Adaptive filtering schemes using a least mean squares (lms) and a normalized least mean squares (nlms) approach were employed to extract lung sounds from the corrupted signal. Up to 25 dB of noise reduction over the 100–600 Hz frequency range was achieved with the lms algorithm, with the more complex nlms algorithm providing faster convergence and up to 5 dB of additional noise reduction. These findings indicate that a combination of active and passive noise reduction can be used to measure clinically useful lung sounds in high noise environments. [Work supported by US Air Force.]

2:30

2pBV5. Echo-based neonatal breathing tube monitoring system. Jeffrey P. Mansfield, George R. Wodicka (School of Elec. and Comput. Eng. and Hillenbrand Biomedical Eng. Ctr., Purdue Univ., West Lafayette, IN 47907-1285), and Daniel C. Shannon (Massachusetts General Hospital, Boston, MA 02114)

A device to monitor the position and patency of neonatal breathing tubes (endotracheal tubes, ETT) was developed based on a pulse-echo technique originally employed in a system for adult breathing tubes [J. P. Mansfield and G. R. Wodicka, J. Sound Vib. **188**(2), 167–188 (1995)]. A 100 μs duration sound pulse is emitted into the neonatal ETT via a wave tube containing a miniature sound source and a miniature microphone. The microphone detects echoes from acoustic impedance changes arising largely from changes in cross-sectional area within the ETT lumen and airways. The delays and amplitudes of echoes from obstructions within the ETT are used to estimate their respective locations and sizes. An echo arising from the ETT tip provides an estimate of the cross-sectional area just beyond the tube tip and is used to distinguish between proper tracheal and erroneous bronchial or esophageal intubations. Determination of tracheal position is accomplished by tracking the delay of a characteristic echo which arises from the airways. During intubation in five anesthetized rabbits, the device tracked changes in ETT position with an accuracy of 1 ± 1.7 mm (mean$\pm$s.d.) over 188 measurements, which is adequate for use in even the smallest infant.

2:45

2pBV6. Spatial information in simultaneous multimicrophone recordings of thoracic sounds. Martin Kompis and George R. Wodicka (School of Elec. and Comput. Eng., Purdue Univ., West Lafayette, IN 47907-1285)

An algorithm for the reconstruction of the distribution of sound sources within the thorax was developed. For any point within the thorax, the algorithm tests the hypothesis that it contains the only relevant source to explain a given set of simultaneous measurements on the chest wall. A hypothetical source signal is calculated by a least-squares estimation procedure for all points 1 cm apart on a equally spaced grid overlying the thoracic volume. The explained fraction of the total signal variance of all microphones is then displayed as a function of space in a 3-dimensional grey-scale coded image. The algorithm was applied to simultaneous lung sound recordings with eight microphones from two healthy male subjects. Preliminary findings suggest that inspiratory sounds are produced more peripherally in the thorax, while expiratory sounds originate more centrally. Some of the images depict features which correlate with anatomical structures such as the large airways, the heart, or during the process of swallowing, the esophagus. A mechanical lung model is being developed to investigate the significance and reproducibility of these images features. [Work supported by Swiss National Research Foundation, Roche Research Foundation, and NSF.]

3:00

2pBV7. A modal characterization of the ovine head. Teresa A. Whitman, Mark T. Morgan (School of Agricultural and Biological Eng., Purdue Univ., West Lafayette, IN 47907-1146), and George R. Wodicka (Purdue Univ., West Lafayette, IN 47907-1285)

Although the vibrational response of the head has been observed to correlate with intracranial pressure when measurements are made directly on the skull [M. Stevanovic *et al.*, Ann. Biomed. Eng. **23**, 720–727 (1995)], the effect of the extracranial tissues of an intact head has only been hypothesized. In this study, the modal characteristics of the freely vibrating ovine head under various stages of intra- and extracranial tissue removal were determined. Frequency responses from 20 to 2000 Hz were measured using an accelerometer and impact hammer at 32 sites on 5 intact ovine heads, and after progressive removal of the jaw, intra- and extracranial tissues, and finally on the boiled and dried skull. Intra- and extracranial tissues increased modal damping and decreased natural frequencies. Dried skulls exhibited as many as 5 lightly damped resonances with the lowest occurring around 1 kHz, whereas only 2 heavily damped resonances were evident prior to boiling. The first resonance for the heads at various stages of tissue removal occurred between 200 and 400 Hz and similar mode shapes between heads were found at this lowest resonance as compared by the Modal Assurance Criterion. As natural frequencies increased, the modes became more complex, resulting in dissimilar mode shapes between heads. [Work supported by NSF.]

3:15

2pBV8. Measurements of the effect of polypropylene vials on ultrasound propagation. Robin O. Cleveland, Michalakis A. Averkiou, Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), and James A. McAteer (Indiana Univ. School of Medicine, Indianapolis, IN 46202-5120)

Polypropylene vials are commonly used in *in vitro* experiments to hold cell preparations that are exposed to ultrasound. The vial has an acoustic impedance very close to water, in which case there should be little transmission loss as sound propagates into the vial. Measurements of the acoustic field within polypropylene vials have been carried out using both pulsed medical ultrasound and lithotripter shock waves. It has been found that for certain orientations of the acoustic field and vial there is significant loss of pressure amplitude within the vial. In particular, sound that enters the vial through the round, hemispherical end is attenuated and distorted. Hot and cold spots within the vial are also measured. When the rounded end is replaced with a flat surface very little distortion and attenuation of the waveform occurs. The distortion induced by the round end is apparently due to refraction introduced by the vial—the speed of sound in polypropylene is about 1.7 times that of water. A simple ray analysis indicates the presence of hot and cold spots within the vial in qualitative agreement with observations. These results indicate that cell injury experiments may be dependent on the vial orientation. [Work supported by NIH and the ASA Hunt Fellowship.]

Session 2pEA

Engineering Acoustics: Surface Acoustic Wave (SAW) Devices

Harold C. Robinson, Chair
Code 2131, Naval Undersea Warfare Center, New London Detachment, New London, Connecticut 06320

Invited Papers

2:00

2pEA1. Types and properties of surface acoustic waves. Harold C. Robinson (Naval Undersea Warfare Ctr., New London, CT 06320)

Surface acoustic waves (SAWs) are used for a wide variety of signal processing and sensing applications at high frequencies. SAW devices are attractive for these applications because of their small size and low cost relative to many competing technologies. However, SAWs are physically quite different from the acoustic waves encountered in sonar and medical ultrasonics. In addition, the family of surface acoustic waves is quite diverse, with each type of wave having unique electrical and mechanical characteristics. This presentation will describe the physical properties of surface acoustic waves and delineate what distinguishes SAWs from other types of elastic waves. The simple example of an isotropic elastic material will be examined in some detail to illustrate many of the fundamental concepts of SAWs. Methods of generating and detecting SAWs used in device applications will be shown. A simple interdigital SAW device will be described in terms of its composition, design, and electrical and mechanical properties. A brief historical overview on the development of surface acoustic wave devices will also be presented.

2:30

2pEA2. Overview of SAW materials. Arthur Ballato (US Army Res. Lab., AMSRL-PS, Fort Monmouth, NJ 07703-5601)

There is now a rich variety of piezoelectric materials available for surface acoustic wave applications. These include crystals in symmetry classes $mm2$, 32, $3m$, $4mm$, $6mm$, and $4\bar{a}3m$. Classes $6mm$ and $4\bar{a}3m$ include most of the binary semiconductors, such as gallium arsenide and related alloys, making integrated acoustic/electronic devices possible. Depending upon use, the SAW substrate material may be chosen to have certain combinations: piezocoupling, temperature coefficient of frequency or delay, small coupling to bulk modes, low sensitivity to acceleration or thermal transients, high acoustic Q, small power flow angle, etc. In this paper, these and other criteria are discussed broadly, with illustrations of pertinent materials and orientations. Also given are recommendations of substances for which the material constants are well enough known to be practical, as well as indications for necessary future work in growth and characterization. A broadband equivalent electrical network for SAW propagation is presented, by means of which the appropriate material parameters may be extracted from network analyzer measurements. The circuit can also serve to represent the salient features of devices in their working environments.

3:00

2pEA3. Surface acoustic wave device fabrication. Donald C. Malocha (Elec. and Comput. Eng. Dept., Rm. 407, Univ. of Central Florida, Orlando, FL 32816-2450)

Many frequency control operations for wireless and satellite communication systems are provided by acoustic solid-state devices in the operating frequency range from 10 MHz to 3 GHz. Surface acoustic wave (SAW) device technology is a key technology because it is solid state, monolithic, high performance, and low cost. An important parameter is the versatility and simplicity in device fabrication since many techniques developed by the integrated circuit industry can be applied.

3:30–3:45 Break

3:45

2pEA4. Surface acoustic wave microsensors. John F. Vetelino (Lab. for Surface Sci. and Technol., Sawyer Res. Bldg., Univ. of Maine, Orono, ME 04469)

Surface acoustic waves (SAWs) excited on the surface of piezoelectric crystals form the basis of a family of microsensors that is sensitive, portable, cheap, and small. These microsensors can be used in a wide variety of applications to sense low levels of gases, physical quantities such as mass, temperature, pressure, flow and humidity, biological processes, corrosion, ions and particulate matter, and a variety of mechanical and electrical quantities. In contrast to other microsensor technologies, SAW microsensors can sensitively detect both electrical and mechanical phenomena. This is due to the fact that the SAW has both mechanical and electrical fields. In those applications requiring sensitivity, such as gas sensing and biosensing, the selectivity is provided by a receptor film applied to the piezoelectric substrate. In this paper different types of SAWs are described and prototype sensing geometries are presented. Particular attention is devoted to one of the most common SAW microsensor applications, namely, gas sensing. Finally, results obtained using SAW microsensors in a variety of applications are presented and discussed.

4:15

2pEA5. A review of progress in acoustic charge transport technology. William D. Hunt (School of Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0250)

Acoustic charge transport (ACT) devices are essentially charge coupled devices for which the potential propagating with a surface acoustic wave (SAW) in a piezoelectric semiconductor traps and transports charge. In this paper a physical description of device operation for which both acoustic and electronic attributes play a key role is given. Next, the various acoustic and electronic modeling tools that have been developed to aid in the design of ACT devices will be discussed. There have been several architectures which have been reported to date and the virtues of these approaches will be compared. There will be a discussion of some of the problems that have kept the technology from gaining widespread acceptance and some predictions of the areas of research which will be the most promising.

TUESDAY AFTERNOON, 14 MAY 1996 — REGENCY A, 1:00 TO 2:00 P.M.

Session 2pMUa

Musical Acoustics: Distinguished Lecture

Uwe J. Hansen, Chair

Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Invited Paper

1:00

2pMUa1. Nonlinearity and the sounds of musical instruments. Neville H. Fletcher (Australian Natl. Univ., Canberra, Australia)

Musical instruments can be classified into two groups—those that have a continuous energy input and produce a sustained sound, and those that have an impulsive energy input and produce a transient decaying sound. Instruments of the first group, which includes violins, flutes, clarinets, trumpets, etc., can be described as "essentially nonlinear," by which it is meant that their musical operation depends critically upon the presence of nonlinearity, and without it they would scarcely sound at all. The transient or percussive group of instruments, which includes pianos, guitars, bells, gongs, and cymbals, on the other hand, can be described as "incidentally nonlinear" implying that, while nonlinearity may influence their sound, its presence is by no means essential. Paradoxically, inadequate nonlinearity in a sustained-tone instrument leads to complex behavior, while complexity in a transient instrument is enhanced by strong nonlinearity! This talk explores some of these matters in a simple physical way, with demonstrations from particularly interesting instrument types.

Session 2pMUb

Musical Acoustics: General Topics: Mostly Perception

Ian M. Lindevald, Chair
Division of Science, Northeast Missouri State University, Barnett Hall, Kirksville, Missouri 63501

Contributed Papers

2:10

2pMUb1. Pitch perception of spectrally dense musical chords in a highly reverberant church. Ian M. Lindevald (Div. of Sci., Northeast Missouri State Univ., Kirksville, MO 63501)

Pitch changes perceived across the transition from a driven room sound field to a freely decaying room sound field were studied in a subjective experiment. Spectrally dense organ chords were recorded in a church and later gated to provide samples from both the driven and the decaying sound fields. Terhardt's dynamic pitch algorithm was used to analyze the chord samples for comparison with experimental results. On average, when choosing a lower octave sinusoid for tuning, subjects tuned flatter than when choosing a higher octave. This result is consistent with the predictions of Terhardt's algorithm. Second, there was a consistent trend to perceive the reverberant sound field an average of about 20 cents flatter than the corresponding driven sound field. This effect is not predicted by Terhardt's pitch model. [Work supported by the ASA's Hunt postdoctoral fellowship of 1987–88.]

2:25

2pMUb2. Integrality of first inversion C-major chord components. Barbara E. Acker and Richard E. Pastore (Dept. of Psych., SUNY, Binghamton, NY 13850)

Previous work [Acker and Pastore, Percept. Psychophys. (in press)] using a discrimination version of the Garner paradigm found the E and G frequencies of a root position C-major chord to be integral, with both frequencies exhibiting significant redundancy gains in the correlated condition, but with the E exhibiting less interference in the orthogonal condition than the G. This finding raised the question of whether attention was better allocated to the E frequency specifically, or if it was a function of the middle spectral location. The current work addresses this question by evaluating discrimination of chord components in a first inversion chord, where the E is the lowest frequency component instead of the middle component. Subjects completed fixed, correlated, and orthogonal discrimination tasks for each frequency. Replicating the earlier results, both frequencies experienced significant redundancy gains in the correlated condition, and the E frequency again exhibited less interference in the orthogonal condition. These results will be discussed in terms of attention being more easily allocated to the E frequency specifically, regardless of spectral location. [Work supported by AFSOR.]

2:40

2pMUb3. Nonlinear dynamics of rhythm perception in performed music. Edward W. Large (Inst. for Res. in Cognit. Sci., Univ. of Pennsylvania, 3401 Walnut St., Philadelphia, PA 19104-6228) and Caroline Palmer (Ohio State Univ., Columbus, OH 43210)

Listeners are readily able to identify and track nearly periodic components in complex auditory rhythms such as music, despite the large temporal modulations that commonly exist in music performance. The modulations that performers introduce, however, often communicate musical information such as melody, phrase structure, and meter. A dynamical system model of beat tracking in music performance is first described, in which a performed (external) rhythm serves as driver, and an internal "attentional rhythm" is modeled as a nonlinearly driven oscillator. Next, a coupled two-oscillator network, designed to simulate the perception of multiple beat periods (meter), is introduced. The one- and two-oscillator models' ability to track beats and recover meter is tested in piano performances of Bach's Two-Part Inventions. The coupled oscillator network not only extends the model to the perception of meter, but also influences the ability of each component oscillator to track a component periodicity in the external rhythm. The ability of the network to recover other types of musical information, such as phrase structure, from performance timing is discussed. [Work supported by NSF SBR-8920230 and NIMH R29-MH45764.]

2:55

2pMUb4. Musical stability and melodic implication. William Forde Thompson and Murray Stainton (Atkinson College, York Univ., ON, Canada)

Five perceptual principles of melodic implication, derived from Narmour's (1990, 1992) "implication-realization" theory of melody, were evaluated in an analysis of European folk songs. The principles apply only to unstable or unclosed melodic conditions, and define how such conditions initiate processes of pattern recognition which then generate expectations about the properties of subsequent notes. For instance, the principle of proximity states that unstable events generate an expectation that the next note will be proximal in pitch. Musically stable and unstable events were identified in over 700 melodies, using criteria based on metric stress, tonality, and duration. For each event, the next note (the continuation note) was recorded. Multinomial loglinear analysis assessed whether the five principles, separately or in combination, could predict the frequency with which different continuation notes followed different conditions of stability. For continuation notes following unstable events, statistical support was found for all principles. The link between musical stability, melodic expectation, and composition is discussed.

3:10

2pMUb5. Instrument-specific constraints on musical performance: Playing the piano and the guitar. Timothy M. Walker (Dept. of Psych., Ohio State Univ., Neil Ave., Columbus, OH 43201)

Kendall and Carterette (1990) propose a chain of musical communication similar to linguistic communication, whereby composers, performers, and listeners all share some amount of musical knowledge. Although the authors emphasize the extent to which information is communicated, there are also important transformations which take place. The physical constraints of an instrument may play a significant role in the interpretive process by operating as musical filters. Improvisation is considered as a task in which all three links in the communicative chain (composer, performer. and listener) are simultaneously represented by a single musician. An experiment was conducted in which participants were asked to perform improvisations on a guitar or piano, each of which functioned as an MIDI instrument. The sounded timbre was independently varied, to produce either a guitar or a piano sound. Participants displayed different patterns of key choice and note usage depending upon which instrument they were playing, suggesting that the physical constraints of the instrument influ-

enced their performance. Participants also employed expressive guitar pitch bends only in the guitar timbre, suggesting that higher-level knowledge of instrument characteristics, as well as external physical constraints, can influence musical performance. [Work supported by NIMH grant R29-MH45764.]

3:25

2pMUb6. Rate change effects on the performance of musical sequences. Rosalee K. Meyer, Caroline Palmer (Ohio State Univ., Columbus, OH 43210), and Leonard B. Meyer (Univ. of Pennsylvania)

Changes in presentation rate can cause perceptual reorganization of rhythmic structure in music. This finding is extended to music performance. One music-theoretic explanation suggests that larger-scale structures such as meter are perceptually prominent at fast rates and smaller-scale structure such as segmentation of event sequences into groups are perceptually prominent at slow rates. In contrast, theories of motor programming predict that event durations during performance should remain proportionally constant across rate changes (relational invariance). In an initial study, pianists performed the same musical sequence at different rates with different grouping structures. Interactions of grouping and sequence rate on component note durations support the music-theoretic explanation of perceptual reorganization and suggest why relational invariance fails in complex behaviors. However, the role of tactus (perceptually prominent beat level) was not considered in this study; perception of tactus differs across presentation rates. The interaction of tactus with rate changes in perceptual reorganization of musical structure is investigated in a performance task. It is predicted that higher tactus levels (longer beat durations) are emphasized (more accurate, less variable) at fast rates and lower tactus levels (shorter durations) are emphasized at slow rates. [Work supported by NIMH R29-MH45764.]

3:40

2pMUb7. Mothers and their children hear a musical illusion in strikingly similar ways. Diana Deutsch (Dept. of Psych., Univ. of California, San Diego, La Jolla, CA 92093)

A two-tone pattern, known as the tritone paradox, has certain curious properties. It is heard as ascending when played in one key, yet as descending when played in a different way. Further, when the pattern is played in any one key it is heard as ascending by some listeners but as descending by others. Perception of this pattern has been found to vary in correlation with the geographical region in which the listener grew up, and also with the pitch range of the listener's speaking voice [D. Deutsch, Philos. Trans. R. Soc. London Ser. B **336**, 391–397 (1992)]. A striking correlation is here reported between how a listener hears this pattern and how his or her mother hears it. Fifteen subjects (ten children and five adults) were studied. The subjects were all Californians; however their mothers had grown up in diverse geographical regions, including England, the European continent, and various parts of the United States. The perceptions of all the subjects corresponded more closely to those of their respective mothers than to the average of all the mothers; this effect was highly significant statistically. Implications of these findings are discussed.

3:55

2pMUb8. Directed attention and perception of frequency changes. Barbara E. Acker and Richard E. Pastore (Dept. of Psych., SUNY, Binghamton, NY 13902-6000)

What role does attention play in the perception of components within a melody? A paradigm used by Palmer and Holleran [Percept. Psychophys. **56**(3), 301–312] provides a basis for addressing this question. Their study investigated different influences on the perception of pitch alterations in three-voiced musical passages, where subjects first learned a four-measure standard, then completed a same/different task. "Different" passages were constructed by altering the original standard, with a harmonically related (HR) or unrelated (HU) frequency change made in the lowest, middle, or upper voices of the musical passage. Results indicated that HR changes were less detectable than HU changes and were the least detectable in the middle voice. The current study uses the same procedures and musical materials as the Palmer and Holleran work, but provides the subjects with cues indicating which frequency region might contain the alteration (low, medium, or high). Results will be discussed in terms of the role directed attention plays as a function of harmonic relationship and frequency region.

4:10

2pMUb9. Design and measurements of variably nonuniform acoustic resonators. Bruce Denardo and Miguel Bernard (Dept. of Phys. and Astronomy, and Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The design, construction, and acoustical measurements of resonators with nonuniform cross-sectional areas that are easily altered to yield different resonance frequencies are described. These resonators are useful as educational demonstrations of symmetry breaking and of an effect of nonuniformity upon standing waves, and have been previously demonstrated and explained [B. Denardo, J. Acoust. Soc. Am. **95**, 2935(A) (1994), and B. Denardo and S. Alkov, Am. J. Phys. **62**, 315–321 (1994)]. Resonators that yield two, three, and four pitches are considered, where the relative frequencies are designed to correspond to musical intervals. Agreement is within 2% in all cases and 1% for most. The data reveal a breakdown of the theory, which is shown to be a result of additional kinetic energy, and thus effective inertia, near a discontinuity in cross-sectional area. The data also reveal that an appropriate end correction for this case is not that for a thin-walled tube but, rather, an infinite flange.

Session 2pMUc

Musical Acoustics: The Ohio State University Wind Ensemble

James M. Pyne, Chair
School of Music, Center for Cognitive Science, The Ohio State University, 1866 College Road, Columbus, Ohio 43210

Chair's Introduction—5:15

Invited Paper

5:20

2pMUc1. The Ohio State University Wind Ensemble. Gary Lewis and Kathleen Gardiner (School of Music, Ohio State Univ., 1866 College Rd., Columbus, OH 43210)

The Ohio State University Wind Ensemble musicians are the finest wind and percussion performers, both graduate and undergraduate, at the Ohio State University. These gifted young players will present an exciting program that displays their remarkable range of musical and dramatic capability. Youthful and dynamic, conductor Gary Lewis is in his premiere season as OSU director of bands. He comes to OSU from the University of Michigan, where he served as director of the Michigan Marching Band and as conductor of the Wind Ensemble and Contemporary Directions Ensemble. Professor Lewis is active as a guest conductor throughout the United States and is highly respected in the realm of both wind ensemble and orchestral conducting. Clarinet soloist Kathleen Gardiner, holds a prestigious Research and Graduate Council Fellowship and is a doctoral candidate at OSU. She was the first place winner of the International Clarinet Association solo competition in 1994 and the Eastman School of Music concerto competition in 1993. As a professional performer Ms. Gardiner serves as principal clarinet of the Pro Musica Chamber Orchestra.

TUESDAY AFTERNOON, 14 MAY 1996 MT. RUSHMORE, 1:00 TO 3:35 P.M.

Session 2pNS

Noise, Architectural Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Curt Holmer Retrospective

David Lubman, Cochair
David Lubman and Associates, 14301 Middletown Lane, Westminster, California 92683

Richard H. Lyon, Cochair
RH Lyon Corp, 691 Concord Avenue, Cambridge, Massachusetts 02138

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. A consultant for acoustical product breakthroughs—A tribute. John W. Kopec (Riverbank Acoust. Labs. (RAL™), 1512 Batavia Ave., Geneva, IL 60134)

The staff at the Riverbank Acoustical Laboratories (RAL™) in Geneva, IL is often asked to recommend an acoustical consultant for a myriad of acoustical related projects. In order to avoid any unfavorable backlash situations, the staff has to be as certain as possible that they recommend a consultant who will successfully fulfill the request at hand. To assist in the selection process a three-level-type list of tasks, including the names of various acoustical consultants that might fulfill these tasks, was compiled. Level three included the names of acoustical consultants that possibly could fulfill, perhaps, one of the more demanding kinds of requests, which was the need for an acoustical consultant that could provide the design of an acoustical breakthrough product. Curt Holmer fulfilled that request three times, and because of that he had earned a well-deserved place on the list of outstanding acoustical consulting accomplishments that at the time of their occurrence provided an acoustical breakthrough.

1:30

2pNS2. Curt Holmer in retrospect. G. Maidanik (David Taylor Model Basin, NSWC, Bethesda, MD 20084-5000)

When Curt joined BBN I had already left, nonetheless at ASA and other societies' meetings we met and discussed many scientific issues, past and present. His interest in statistical energy analysis (SEA) was particularly intense, but not to the exclusion of many other interesting topics in the analysis of structural acoustics and noise control. He was a patient listener and a critic of note, but always gentle and empathetic. I enjoyed and valued his company and, as many others who knew him, will continue to miss his presence.

1:55

2pNS3. "Why don't we make the membrane loose?" Twelve years of noise control foams. J. Stuart Bolton (1077 Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-1077)

Curt Holmer was well known to me while I was a graduate student in England through the chapter in *Noise and Vibration Control* that he co-wrote with Istvan Ver. In a book that may be the most commonly referred to on acoustics, his was the chapter that I found most useful for both its qualitative and quantitative insights. When I came to Purdue in 1984 it was a pleasant surprise to find both that Curt was working nearby in Indianapolis and that he had an interest in noise control foams of the type that I had studied during my Ph.D. I soon found an opportunity to invite myself down to meet him at EAR, and, like many others before, discovered him to be a gracious and generous host. We became good friends, and over the next decade he actively encouraged my interest in both theoretical and practical aspects of noise control foams (even though when we first met he gently revealed that EAR was already selling a product based on what I had thought was my own "revolutionary" insight: see title). In this presentation, progress in the area of foam modeling over the time that I knew Curt will be reviewed.

2:20

2pNS4. The acoustics of structures with fuzzy joints. Richard H. Lyon (RH Lyon Corp., Cambridge, MA 02138)

Curt Holmer was always interested in the practical applications of statistical energy analysis, and the author had many discussions with him regarding its uses, and areas of transition to deterministic analysis. In some of the latest discussions, the topic of the "mid-frequency range" dominated, referred to by the author as "the awkward zone." In this transition range between deterministic and fully probabilistic models, the greatest uncertainty may be in the joints—their actual location, and their rigidity. This suggests a hybrid approach, in which the substructures are treated as deterministic, and the joint locations and rigidities are uncertain, or in popular terms, fuzzy. This approach is exemplified by the joining of two plates by a link having a location and stiffness that are variable over a small range. The results of the analysis support such an approach, in that they show the kind of transition that has been noted in the awkward zone.

2:45

2pNS5. Evaluation of a technique to identify models of modes in nonlinear structures. S. A. McCabe and P. Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-1077)

Interactions between modes in nonlinear structures can lead to situations where, for the same excitation, different responses may be observed. The observed response will be a function of the system, of the excitation, of initial conditions, and of the effects of small perturbations to the system. Some responses may be observed more often than others. When the more rarely observed responses are perceived as problem situations, it is often frustrating to demonstrate that the problem exists, because it is difficult to control the test sufficiently to repeatedly observe the behavior. Analysis of system models can be used to identify under what conditions these responses may occur, and indicate how the region where such responses are possible may be approached. In real systems, the model structure for the system has to be identified, which may be an affirmation of a theoretical model structure, and the parameters of these models estimated. Described in the presentation are the results of using a system identification technique to construct models relating the excitation to the response of two interacting modes in a thin plate under tension.

3:10

2pNS6. High frequency modeling of vehicle noise performance of aluminum body structure. James J. Lee, Curt Holmer, and Mark J. Moeller (Ford Motor Co., Adv. Eng. Ctr., Dearborn, MI 48121)

This presentation is dedicated to Curt Holmer's contribution to acoustics at Ford Motor Co. The presentation material is based on the work he was involved with during the last part of his life. The results of the 1992 Taurus SEA model are used to assess the effect of an aluminum body structure. The noise performance comparison includes airborne and structure-borne paths. The effect of structural damping is also simulated.

Session 2pPAa

Physical Acoustics and Bioresponse to Vibration and to Ultrasound: Workshop on Therapeutic Applications of Medical Ultrasound II

Jeffrey B. Fowlkes, Chair
Department of Radiology, University of Michigan Medical Center, Kresge III R3315, Box 0553, Ann Arbor, Michigan 48109

This Workshop is an attempt to bring a "small meeting" environment into a big meeting. In addition to the tutorial and special presentations outlined below and in the related sessions (2aPAa, 3aPAa); there will be significant participation by other workshop attendees; discussion among participants is a major objective of this program. Therefore, a specific timetable will not be followed during the workshop.

Registration for the tour to Indiana University School of Medicine to be held on Wednesday afternoon (see Session 3pPA, page 2527) will be held at this session. Space on the tour is limited.

Chair's Introduction—1:00

Invited Papers

2pPAa1. The demographics of cavitation produced by medical ultrasound. Ronald A. Roy (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

A frequent consequence of moderate-to-high intensity ultrasound is the nucleation and proliferation of acoustic cavitation activity. Nucleation refers to the formation of a gas/vapor bubble resulting from acoustic stress. Cavitation activity is the subsequent energetic motion of these bubbles, and the associated mechanical and thermal effects. The process is very nonlinear. It is appropriate to view the family of cavitation as occupied by distinct, yet related phenomena. For example, if one drives a bubble at moderate amplitudes, stable motion results accompanied by fluid microstreaming. If the same bubble is driven hard, explosive growth is followed by inertial collapse, together with the formation of shock waves, free radicals, etc. When assessing the likelihood of a mechanical bioeffect, one must consider the characteristics of the cavitation field, which can be estimated knowing the properties of the medium, and the properties of the sound field. A tutorial description of basic cavitation dynamics is presented. Attention is given to those aspects of bubble motion that are relevant to medical ultrasound conditions. Associated mechanical and thermal effects are discussed, and the relationships between the characteristics of the sound field and the expected cavitational response of the medium explored. [Work supported by NIH Grant PO1 DK43881-01.]

2pPAa2. Resonant bubbles and bioeffects mechanisms. Edwin Carstensen (Rochester Ctr. for Biomedical Ultrasound, Univ. of Rochester, Rochester, NY 14627)

2pPAa3. Potential applications of acoustic generation of arterial microbubbles. P. L. Carson, J. B. Fowlkes, E. A. Gardner,[a] M. Bruce, and J. M. Rubin (Dept. of Radiology, Univ. of Michigan Medical Ctr., Ann Arbor, MI 48109-0553)

With intense, focused ultrasound pulses it is possible to generate microbubble boluses in the arterial system with potential diagnostic and therapeutic uses. Arterial microbubbles have been generated recently in flowing whole blood *in vitro* with 725 kHz ultrasound passing through canine tissues simulating human transcutaneous generation. Similar generation was reported previously [J. A. Ivey *et al.*, Ultrasound Med. Biol. **21**, 757–767 (1995)] from interoperative, *in vivo* exposures at 1.8 MHz and at higher intensities. If biological effects can be controlled acceptably, knowledge of ultrasonic bioeffects will be enhanced. Short boluses of ultrasonically imageable, less than 40 micron bubbles, generated in selected arteries, might be usable for diagnosis and monitoring of those macrovascular and perfusion abnormalities which currently are evaluated more slowly and, presumably, expensively. Very sparse distributions of 30 to 40 micron bubbles should provide point beacons with which to refocus ultrasound beams for high resolution imaging and treatment through aberrating tissues including the skull. Identification of feeder arteries to and disruption of flow to therapeutic targets should be achievable to enhance chemical or ultrasonic therapy. [Work supported in part by USPHS grant No. 2RO1 DK 42290 from the Nat'l Inst. Diab., Digestive and Kidney Disease.] [a]Current address: Acuson, Inc., Mountain View, CA.

2pPAa4. Intestinal hemorrhage from exposure to pulsed ultrasound and lithotripter fields. Diane Dalecki, Sally Z. Child, Carol H. Raeman, David P. Penney, and Edwin L. Carstensen (Rochester Ctr. for Biomedical Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Using ultrasound with pulse lengths of 10 μs and pulse repetition frequencies of 100 Hz, the thresholds for intestinal hemorrhage in mice after a 5-min exposure range from ~ 1 MPa at 1 MHz to ~ 4 MPa at 4 MHz. With a piezoelectric lithotripter, the threshold for intestinal hemorrhage after 200 pulses was 1–3 MPa. In neither case can the effects be explained by heating. As a test for cavitation, the extent of hemorrhage in the gas-containing intestines of pregnant mice was compared to the amount of hemorrhage in the gas-free intestines of their fetuses. On the 18th day of gestation, the abdominal regions of pregnant mice ($n=6$) were exposed to 200 pulses from a piezoelectric lithotripter. Acoustic pulses had a peak pressure amplitude of 10 MPa and were administered at a rate of ~ 1 Hz. All maternal intestines showed hemorrhagic regions extending several centimeters in length. In contrast, only 1 of 43 exposed fetuses showed an intestinal hemorrhage and this one lesion was less than one millimeter in diameter. These are consistent with a cavitation-related mechanism for the production of intestinal hemorrhage by exposure to acoustic fields.

2pPAa5. Effects of ultrasound on function of the amphibian and the mammalian heart. Diane Dalecki, Alan MacRobbie, Sally Z. Child, Carol H. Raeman, and Edwin L. Carstensen (Rochester Ctr. for Biomedical Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Exposure to the frog heart to single, millisecond length pulses of ultrasound has been shown to cause effects by two qualitatively different physical mechanisms. One of these effects has also been observed in mice. When the pulse is delivered during systole, it can cause a reduction in the developed aortic pressure. Evidence suggests that this arises from an action of radiation force on the heart tissues. The effect can be produced with an acoustic reflector over the heart. This maximizes the radiation force delivered to the heart, but eliminates direct interaction of the ultrasound with the heart tissue, thus experimentally eliminated heating and cavitation as mechanisms of action. When the pulse is delivered during diastole, it can produce a premature ventricular contraction. No premature ventricular contractions were observed with the acoustic reflector over the heart. The mechanism is nonthermal. Available evidence does not preclude cavitation as the primary physical mechanism responsible for premature ventricular contractions.

2pPAa6. Contrast agents enhance biological effects *in vivo*. Diane Dalecki, Sally Z. Child, Carol H. Raeman, Charles W. Francis, David P. Penney, and Edwin L. Carstensen (Rochester Ctr. for Biomedical Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Hemolysis *in vivo* was demonstrated by exposing the hearts of mice to focused, pulsed ultrasound (1-MHz carrier, 10-μs pulses, 100-Hz repetition frequency, 5-min total exposure time) while administering a contrast agent *via* tail vein. The threshold was approximately 4 MPa. This study suggests that inertial cavitation can take place *in vivo* if appropriate nuclei are present. However, since the degree of hemolysis in the absence of exogenous nuclei is indistinguishable from sham exposed blood with contrast agents, we must conclude that the number of spontaneous nuclei in normal blood *in vivo* is very small. The administration of contrast agents during lithotripsy enhanced hemorrhage in most of the tissues of the body. Thresholds for hemorrhage in the fields of a piezoelectric lithotripter were <2 MPa.

2pPAa7. Acoustic cavitation and bioeffects in the vascular system. J. B. Fowlkes, E. A. Gardner,[a] and P. L. Carson (Univ. of Michigan Medical Ctr., Ann Arbor, MI 48109-0553)

An *in vivo* bolus generation system has provided information concerning acoustic cavitation in blood and associated bioeffects. Microbubble boluses produced in the canine abdominal aorta by tone bursts of 1.8 MHz focused ultrasound were detected in distal arteries using medical ultrasound scanners and a resonant bubble detector. At the maximum acoustic intensity, 19,000 W/cm^2(I_{SPTA}), 125–250 ms bursts generated contrast enhancement persisting several cardiac cycles. Intensities down to 3,700 W/cm^2 yielded progressively weaker boluses while shorter and weaker boluses were produced with 64–0.5 ms bursts at the maximum acoustic intensity. There was a 50% occurrence of vascular wall effects (by histology) but no immediate mechanical failure. Oxygen ventilation has been shown to destabilize IV injected contrast agents and similarly, cavitation thresholds were elevated in the vena cava. Resonant bubble dectection indicated the presence of 7 μm diameter bubbles but the bubble size range is unknown. These results provide insight into bubble production by acoustic cavitation but any bioeffects must be examined in the context of arterial bolus production where the risk of significant damage may be less than that of alternative procedures. [Work supported in part by USPHS 5 RO1 DK42290.] [a]Current address: Acuson, Inc., Sunnyvale, CA.

2pPAa8. Finite volume model of weak shock interaction with lung tissue. S. M. Gracewski and Zhong Ding (Mech. Eng. Dept. and Rochester Ctr. for Biomedical Ultrasound, Univ. of Rochester, Rochester, NY 14627)

The lung and other organs with high gas content tend to be the most susceptible to damage by lithotripter shock waves. Since the gas volume is large, the damage processes most likely differ from traditional cavitation mechanisms. A finite volume model has previously been developed to investigate the interaction of shock waves with single, isolated bubbles. This model has been modified to explore other possible mechanisms of lung tissue damage. In the model, the single bubble is replaced with a layer of cylindrical bubbles, and a shock wave is incident normal to this bubble layer. The affect of bubble spacing on the shock wave propagation through the layer and the resulting bubble dynamics is determined.

2pPAa9. Bioeffects in rat lungs from diagnostic ultrasound. Christy Holland (Univ. of Cincinnati, College of Medicine, Dept. of Radiology, P.O. Box 67042, Cincinnati, OH 45267) and Robert Apfel (Yale Univ., New Haven, CT 06520-8286)

Session 2pPAb

Physical Acoustics: Porous Media and Wave Propagation

James P. Chambers, Chair
National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677

Contributed Papers

1:15

2pPAb1. Effect of anisotropy on elastic wave propagation in fluid-saturated porous media. Andrei D. Degtyar, Stanislav I. Rokhlin, and Laszlo Adler (Dept. of Industrial, Welding and Systems Eng., Ohio State Univ., 190 W. 19th Ave., Columbus, OH 43210)

On a mesoscale the pore structure in natural rocks may have preferred orientation (texture) which leads to anisotropy of permeability, tortuosity, and shape factor. It is shown that such an anisotropic fluid saturated porous medium supports four different wave types: fast and slow quasilongitudinal and two quasishear waves. These results indicate that the velocities of the fast quasilongitudinal and two quasishear waves mostly depend on the properties of the frame and are not sensitive to the permeability and tortuosity directly (the frame stiffnesses, permeability, and tortuosity are indirectly related due to dependence on pore structure, however formally they can be considered as independent parameters). Thus by measuring these velocities one could determine frame elastic constants. The slow wave velocity, on the contrary, mostly depends on the pore geometry. Its angular dependence in a water- or air-saturated solid allows us to recover the components of the permeability and tortuosity tensors. This approach opens new challenges in determination of such characteristics of porous materials as preferred pore orientation and tortuosity which had been previously inaccessible experimentally.

1:30

2pPAb2. Acoustic properties of porous model media of spherical particles. H. Tavossi and B. R. Tittmann (Dept. of Eng. Sci. and Mechanics, Penn State Univ., University Park, PA 16802)

The through transmission method of longitudinal ultrasonic pulses was used to investigate the collective acoustic properties of a porous media model of cohesionless spherical particles. The results of this investigation showed dispersion and filtering effects for the transmitted acoustic waves. At low frequencies the wave velocity decreased with increase in frequency and particle size. In contrast at high frequencies the wave velocity decreased with increase in particle size, for relatively large particles ($kR \gg 1$). Calculated results from existing theoretical models for porous media by M. A. Biot, were compared to the experimental results. It was found that new modifications of the existing theoretical model for porous media were necessary in order to interpret the data in a quantitative manner.

1:45

2pPAb3. A mixed displacement-pressure formulation for Biot's poroelastic equations. Noureddine Atalla, Raymond Panneton, Patricia Debergue (GAUS, Mech. Eng., Univ. de Sherbrooke, PQ J1K 2R1, Canada), and J.-F. Allard (Laum, Univ. du Maine, Le Mans, France)

Recently, a displacement (u,U) finite element model for the three-dimensional poroelasticity problem has been developed by Panneton and Atalla [J. Acoust. Soc. Am. 98, 2976(A) (1995)]. This model, while accurate, has the disadvantage of requiring cumbersome calculations for large finite element models and spectral analyses. To overcome this difficulty, this paper presents a mixed displacement-pressure (u,P) finite element model. First, the classical Biot–Allard equations are rewritten in terms of the solid phase macroscopic displacement vector and the fluid phase macroscopic pressure. The new coupled equations have the advantage of writing the poroelasticity equations in the form of the coupling between an equivalent elastodynamic equation for the solid phase and an equivalent Helmholtz equation for the fluid phase. In particular, the equivalent fluid model is transparent in the developed equations. Next, the associated variational formulation is presented together with its numerical implementation. Also the acoustic–poroelastic, elastic–poroelastic, and poroelastic–poroelastic coupling conditions are derived. Several examples are presented to show the accuracy and effectiveness of the proposed model and its coupling with elastic and acoustic media. In particular a systematic comparison is made with the (u,U) finite element formulation. [Work supported by Canadair, CRSNG, and FCAR.]

2:00

2pPAb4. Application of Biot's poroelasticity theory for nonlinear acoustics. Dimitri M. Donskoy (Davidson Lab., Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030), Khaldoun Khashanah, and Thomas G. McKee, Jr. (Stevens Inst. of Technol., Hoboken, NJ 07030)

The nonlinear dynamic equations introduced by Biot to model poroelastic media have not been implemented to describe nonlinear acoustic waves in such media. The equations of the semilinear Biot model are revised and a mathematical model depicting the physical nonlinearity is established. A perturbation technique is then applied to find solutions to the nonlinear Biot equations. The computational results are carried out in details for a one-dimensional model in which the contributions of the fast and slow compressional waves into second harmonic wave are analyzed. The correlation between second and third-order Biot coefficients and measurable nonlinear parameters is also presented along with a parameter analysis.

2:15

2pPAb5. An action of seismic p-wave on fluid level in capillary. Vyacheslav S. Averbach and Yury M. Zaslavsky (Inst. of Applied Phys., Russian Acad. Sci., 46 Ulyanov str., Nizhny Novgorod, 603600, Russia)

It is a desolved problem about fluid level in capillary in homogeneous elastic medium on which the seismic p-wave acts. This wave propagates perpendicular to axis of a capillary. In a linear approach the meniscus vibrates with frequency ω and amplitude as the same one as a classic oscillator. In the next approximation there is the double frequency oscillation and a direct shift of the meniscus. Thus the fluid level will differ from the static one. After averaging the described solution on the orientations and sizes of capillaries the fluid level is estimated in partly saturated porous medium.

2:30

2pPAb6. Dispersion relation influence on rise times of sonic boom propagation through turbulence. Allan D. Pierce (Boston Univ., Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215)

Recent work suggests the "average" turbulence contribution to rise times is accounted for by an extra term in the propagation equation, evolving from an extra term in the dispersion relation $k=(\omega/c)+F(\omega)$, where c is spatially averaged, and, for mechanical turbulence, $F(\omega)$ depends on turbulent energy dissipation rate ϵ per unit fluid mass. Previous analysis suggests that $F(\omega)$ is $-7.48\ C_K e^{i\pi/3}\epsilon^{2/3}c^{-7/3}\omega^{1/3}$. The extra term's influence is explored with a pulse beginning with a stepfunction in pressure which enters a turbulent atmosphere. After propagation through a distance x, the rise time becomes of order of $\epsilon^2 c^{-7}x^3$. This cubic growth is eventually curbed by the nonlinear steepening effect; the above prediction is an upper limit to the turbulence contribution. The discrepancy with the 1971 prediction that the rise time scales as $\epsilon^{4/7}c^{-19/7}x^{11/7}$ is discussed, such being consistent with an $\omega^{7/11}$ term in the dispersion relation. [Work supported by NASA–LRC.]

2:45

2pPAb7. Distorted-wave Born approximation analysis of sound levels in a refractive shadow zone. Kenneth E. Gilbert, Xiao Di, and Rodney R. Korte (Appl. Res. Lab. and the Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Measurements made by Havelock *et al.* show that above a few hundred Hertz, the sound levels in a refractive shadow zone are weakly dependent on frequency. Parabolic equation calculations made recently by Di and Gilbert indicate that the weak frequency dependence is consistent with scattering from small-scale turbulence governed by a Kolmogorov spectrum. In this paper a theoretical analysis is presented using the distorted-wave Born approximation (DWBA) to show why the Kolmogorov spectrum leads to the observed weak frequency dependence. Using DWBA, the problem of scattering by turbulence is formulated in terms of a sampling function that shows the physical origin of the frequency dependence by showing what part of the turbulence spectrum contributes in a typical measurement of levels in a refractive shadow. Results from the DWBA analysis are compared with the data of Havelock *et al.* and with parabolic equation calculations. [Work supported by the Army Research Laboratory and the Applied Research Laboratory.]

3:00–3:15 Break

3:15

2pPAb8. Measurements of wind turbulence spectra near ground and implications for sound propagation calculation. Michael R. Stinson, Gilles A. Daigle, and David I. Havelock (Inst. for Microstructural Sciences, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

Measurements of horizontal wind speed variations have been made at a local airfield using three different techniques and the resulting power spectra compared. The instruments used were a rotating three-cup anemometer, a sonic anemometer, and a hot-wire anemometer. The instrument sampling rates and frequency responses lead to upper frequency limits for the spectra of 0.5, 10, and at least 100 Hz, respectively. Good agreement was obtained between the techniques for those frequencies where they overlap. Knowledge of the shape of the turbulence spectrum is necessary for the incorporation of turbulence scattering into outdoor sound propagation calculations. The observed spectra tend to vary with frequency f as $f^{-5/3}$ over the range 0.001 to 100 Hz (corresponding to turbulence length scales between 5000 and 0.05 m, for a mean wind speed of 5 m/s) although there are significant deviations from this simple dependence. The observed results are consistent with the analysis of Højstrup [J. Atmos. Sci. **39**, 2239–2248 (1982)].

3:30

2pPAb9. Reduction of microphone wind noise using local wind velocity measurements. Michael R. Shust and James C. Rogers (Elec. Eng. Dept., Michigan Technol. Univ., 1400 Townsend Dr., Houghton, MI 49931)

A system for quieting microphone wind noise using local wind velocity measurements is desirable. The concept is shown to be feasible, however, optimization of the cancellation system requires understanding the correlation characteristics between microphone signals (pressure) and anemometer signals (velocity). Wind data was recorded using microphones and anemometers with separation distances from 2.5 to 20 centimeters. The measurements were made in a single plane parallel to the airflow. Both the microphone pressure and wind velocity signals displayed an approximate $1/f$ frequency characteristic. In addition, correlation strength between wind signals (both pressure and velocity) decreased with increasing separation distance. Correlation estimates from microphone to microphone, anemometer to anemometer, and microphone to anemometer are presented. It is demonstrated that adaptive signal processing algorithms can be utilized to reduce microphone wind noise using anemometer signals as a reference.

3:45

2pPAb10. Effects of wind velocity fluctuations on the statistical moments of plane and spherical sound waves in the turbulent atmosphere with the Gaussian correlation function of wind velocity fluctuations. V. E. Ostashev, G. Goedecke (Dept. of Phys., New Mexico State Univ., Las Cruces, NM 88003-8001), and John M. Noble (U.S. Army Res. Lab., White Sands Missile Range, NM 88002)

It is quite often assumed in the theories of waves in random media that medium inhomogeneities have a Gaussian correlation function. Equations for the main statistical moments of plane and spherical sound waves propagating in a medium with temperature fluctuations are well known in the literature. Among these statistical moments are the following: the mean sound field, the variances and correlation functions of log-amplitude and phase fluctuations, and the transverse coherence function. The present paper deals with a derivation of equations for these statistical moments for the case of plane and spherical wave propagation in the atmosphere with wind velocity fluctuations. It is shown that the derived equations can differ not only quantitatively but also qualitatively from analogous equations for the case of sound propagation in the atmosphere with temperature fluctuations even if the wind velocity and temperature fluctuations make the same contributions to the variance of the acoustical refractive-index fluctuations. The results obtained are generalized to the case of sound propagation in an arbitrary medium (for example, for sound propagation in the ocean with current fluctuations). [This material is based upon Work supported by the U.S. Army Research Office under Contract No. DAAH04-95-1-0593.]

4:00

2pPAb11. Acoustic scattering from vegetation. Mark W. Sprague (Phys. Dept., East Carolina Univ., Greenville, NC 27858) and James M. Sabatier (National Ctr. for Physical Acoust., University, MS 38677)

Acoustic scattering from vegetation is a complicated problem that depends on a number of plant characteristics including biomass distribution, plant spacing, plant height, leaf size and shape, and the acoustic impedance of the plant material. A theory for acoustic scattering from vegetation could be the basis for a noninvasive acoustic measurement of some of these characteristics, particularly biomass distribution. Two possible techniques to characterize scattering from vegetation are the multiple scattering approach and the porous medium approach. In the multiple scattering approach, the vegetation is treated as a slab of scatterers with a random distribution. In the porous medium approach, the vegetation is treated as a porous material with a very high porosity. In this talk, both approaches are examined, and their predictions are compared for forms of vegetation such as wheat, corn, cotton, and soybeans.

4:15

2pPAb12. Comparison of sound-speed computation methods. George S. K. Wong (Inst. for Natl. Measurement Standards, National Res. Council, Ottawa, ON K1A 0R6, Canada)

The computation of sound speeds in nitrogen, oxygen, and methane was performed with two methods. Results from the traditional method that relied on virial coefficients B and their temperature derivatives $T\,dB/dT$, $T^2\,d^2B/dT^2$, were compared with those given by a simplified method. At a pressure of 101.325 kPa, and at a temperature of -10 °C, the maximum sound speed deviations between the two methods are within 0.2% and -0.4% for nitrogen and oxygen, respectively. For methane, at the above pressure, and over the temperature range from zero to 450 °C, the maximum deviation is 1.2%. For the above gases, the sound-speed deviations between the methods are largest at low temperatures. It can be shown that by modifying the numerical values of the virial coefficients and their temperature derivatives, the sound-speed deviations between the two methods can be reduced substantially. For nitrogen, when B, $T\,dB/dT$, and $T^2\,d^2B/dT^2$ are modified by multiplication factors of -0.1, 0.55, and 0.7, respectively, the difference between the sound speeds obtained with the two methods decreases to less than ± 0.05%. It is noted that the sign of the numerical value of B is reversed. Uncertainties of the results obtained with both methods were examined.

4:30

2pPAb13. Analysis of electromagnetically induced pressure waves internal to a rigidly confined, electrically conducting layer. John C. Petrykowski (Dept. of Mech. and Aerospace Eng., Univ. of Dayton, 300 College Park, Dayton, OH 45469-0210)

A model is presented that relates how electromagnetically induced pressure waves propagate and grow in response to a steady-state electromagnetic excitation. A normal modes-type solution is used to help account for noncompliant surfaces that bound the layer. In total, three types of wave-like behavior are found: both steady-state and transient acoustic waves and a dispersive wave due to the diffusive nature of the magnetic field. Formulas are presented for the amplitude and phase of each type of wave. The usefulness of these findings for describing acoustic effects in induction systems is discussed.

4:45

2pPAb14. Laser thermo-optical source of sound at a fractal surface in a liquid. Leonid M. Lyamshev (N. N. Andreev Acoust. Inst., Shvernik Str. 4, 117036 Moscow, Russia) and M. L. Lyamshev (Russian Acad. of Sciences, 117242 Moscow, Russia)

Surfaces of real bodies are rough, uneven, or porous as a rule and are characterized by fractal properties. Surfaces may be fractal up to the molecular-level scale. Rough oceanic surface is also fractal. Laser photoacoustic diagnostics of various media have been developing in recent years. It is based on laser excitation of sound (ultrasound or hypersound). Special features of laser sound excitation in a liquid half-space with the rough statistical fractal surface are investigated. It is assumed that an harmonically modulated laser beam is incident on the liquid surface. The height (amplitude) of surface waves is greater than the sound wavelength and the absorption length of light in the liquid. Fractal properties of the surface are characterized by the Weierstrass function and connected with it by a structural function. The Kirchhoff approximation is used to solve the boundary problem. The influence of fractal properties of the surface and parameters of laser radiation and liquid on the amplitude and spatial characteristics of intensity of the acoustic field excited in a liquid by laser radiation is analyzed.

TUESDAY AFTERNOON, 14 MAY 1996 REGENCY B, 1:00 TO 4:45 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Pitch and Loudness

Robert P. Carlyon, Chair

MRC Applied Psychology Unit, 15 Chaucer Street, Cambridge CB2 2EF, England

Contributed Papers

1:00

2pPP1. The effect of mean rate cues on the pitch of filtered pulse trains. Robert P. Carlyon (MRC Appl. Psych. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, England)

Listeners judged which of two bandpass filtered (3900–5400 Hz) pulse trains had the higher pitch. In the control condition, where the two trains differed only in repetition rate, two cues might produce a pitch difference. The interval between any two pulses is a multiple of a common period, which is shorter for the train with the faster rate (the "$F0$ cue"). This stimulus also contains more pulses (the "mean rate cue"). Generally, temporal pitch models focus on the $F0$ cue. The mean rate cue was investigated by, in one condition, deleting a proportion $(1-P)$ of the pulses in each train, and manipulating P so that the mean rate $(P \cdot F0)$ was equal in the two halves of each trial [R. A. Dobie and N. Dillier, Hear. Res. **18**, 41–45 (1985)]. Listeners identified the stimulus having the higher (original) $F0$ as possessing the higher pitch less reliably than in the control condition, and less reliably than in a condition where the mean rate difference was exaggerated. The conclusion that the mean rate cue can influence pitch, was confirmed a second experiment, where listeners adjusted the $F0$s of two pulse trains with different P's to have equal pitches.

1:15

2pPP2. Potential pitch cue for spectral-shape discrimination at high frequencies. Huanping Dai (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

In these experiments on spectral-shape discrimination [D. M. Green, *Profile Analysis* (Oxford U.P., New York, 1988)], the listener's task was to discriminate a standard spectrum in which all components were equal in level, from a signal-plus-standard spectrum in which the level of one component (the signal component) was higher than the level of other components. The threshold was measured as a function of center frequency (250 to 16 000 Hz) for complexes having two or three components. The frequency components were spaced equally on either a logarithmic ($f_{i+1}/f_i = 1.38$) or linear ($\Delta f = 150$ Hz) scale. For all the three-component complexes, in which the signal was added to the center component, discrimination was difficult for frequencies above 2 kHz. For the two-component complexes, discrimination was difficult for the logarithmic but not for the linear spacing. In the latter case, a change in spectral shape caused a change in the center of gravity of the stimulus power spectra, which was perceived by listeners as a change in pitch. The availability of the pitch cue made the discrimination possible for the narrowband two-tone complexes at frequencies up to 8 kHz. [Work supported by NIH.]

1:30

2pPP3. Pitch strength discrimination for iterated rippled noise. William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The pitch strength of iterated rippled noise (IRN) was investigated in a pitch strength discrimination experiment. IRN is generated by a cascade of an add, delay (d in ms), and attentuate (g) circuit, where n represents the number of iterated stages in the circuit. In this study g was negative, which represents the cases in which the noise is subtracted rather than added. IRN produces a pitch, which when n is negative is either ambiguous with a pitch near 10% of the reciprocal of the delay, d, or at one over twice the delay, d. The strength of the pitch varies with g,n, and the type of circuit. Listeners were asked to discriminate between two IRN stimuli each generated with the same d, but with different values of g, and n, and different types of circuits. The discrimination results could only be accounted for by using the first peak in the autocorrelation function of these stimuli. The results will be discussed in terms of recent computational models of pitch processing. [Work supported by NIDCD and AFOSR.]

1:45

2pPP4. A neural computational model for tracking of multiple frequency-modulated tones. Kiyoaki Aikawa and Hideki Kawahara (ATR Human Information Processing Res. Labs., 2-2 Hikaridai, Seikacho, Sorakugun, Kyoto 619-02, Japan)

This paper proposes a novel neural computational model for tracking frequency-modulated (FM) tones. The model can explain various interesting phenomena related to the perception of complex FM tones. The dynamic process in perceiving the pitch of FM tones can be represented by two second-order systems having fast and slow responses [K. Aikawa *et al.*, J. Acoust. Soc. Am. **98**, 2926 (1995)]. An auto-regressive neural matrix model is newly proposed to simultaneously track multiple FM tones. The model is characterized by a novel neural network architecture called the *counter-tonotopic connection* and a new tracking algorithm based on the Lp-norm. Each FM tone is tracked with second-order response characteristics. Several interesting phenomena in perceiving complex FM tones have been reported. For a stimulus tone composed of crossed upward and downward sweep tones, two separate pitch streams were not perceived but a bounced pitch stream was. When both of the crossed sweeps were downward sweeps, two pitch streams were merged and then another new stream was perceived after the crossing point [Matsui, ASJ meeting, 1995-09]. When a sweep tone was followed by white noise, the sweep was perceived as being extended into the noise [Masuda, ASJ meeting, 1995-09]. The proposed neural tracking model successfully replicated these perceived images.

2:00

2pPP5. Information integration of signal frequency. Robert D. Irwin and Daniel L. Weber (Dept. of Psych., Wright State Univ., Dayton, OH 45435)

A listener's ability to integrate information can be evaluated by presenting multiple samples in the distribution discrimination procedure. To examine frequency discrimination with a 2IFC form of this procedure, each interval contains samples drawn from one of two distributions of signal frequency. The listener indicates which interval contained the sample(s) drawn from the distribution with the higher mean. This experiment evaluated the integration of frequency information in terms of improvement in frequency discrimination for an increasing number of samples (n=1,2,3,4,5,6,8,12,16). Seven listeners discriminated a "standard" distribution (means of 400, 565, and 1000 Hz) from each of four "comparison" distributions (means of 401, 403, 406, and 414 Hz for the 400-Hz standard; 566.5, 569.5, 572, and 584 Hz for the 565-Hz standard; and 1002, 1005, 1010, and 1020 Hz for the 1000-Hz standard). All samples were 100-ms sinusoids. All conditions were alike to an ideal observer in that the distributions were normal with a standard deviation equal to the difference between the means. The results indicate that integration of frequency information appears constant across different frequencies when initial performance is equated. [Work supported by AFOSR through WPAFB AL/CFBA.]

2:15

2pPP6. On the relations among temporal integration for loudness, loudness discrimination, and the form of the loudness function. Søren Buus (Communication and Digital Signal Processing Ctr., Dept. of Elec. and Comput. Eng., 409 DA, Northeastern Univ., Boston, MA 02115-5096), Mary Florentine (Northeastern Univ., Boston, MA 02115), and Torben Poulsen (Tech. Univ. of Denmark, DK 2800 Lyngby, Denmark)

Temporal integration for loudness was measured as a function of level from 2 to 60 dB SL using 2-, 10-, 50-, and 250-ms tones at 5 kHz. The adaptive 2I,2AFC procedure converged at the level required to make the variable stimulus just louder than the fixed stimulus. Thus the data yield estimates of the levels required to make tones of different durations equally loud and of the just noticeable differences for loudness level. Results for four listeners with normal hearing show that the amount of temporal integration, defined as the level difference between equally loud short and long tones, varies markedly with level and is largest at moderate levels. The effect of level increases as the duration of the short stimulus decreases and is largest for comparisons between the 2- and 250-ms tones. The loudness-level jnds are also largest at moderate levels and, contrary to traditional jnds for the level of two equal-duration tones, they do not appear to depend on duration. The level dependence of temporal integration and the loudness jnds are consistent with a loudness function [log(loudness) versus SPL] that is flatter at moderate levels than at low and high levels. [Work supported by NIH-NIDCD R01DC02241 and the Technical University of Denmark.]

2:30

2pPP7. Dependence of loudness adaptation on frequency and level. Rhona P. Hellman (Auditory Percept. Lab., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115), Andrzej Miskiewicz (Chopin Acad. of Music, Warsaw, Poland), and Bertram Scharf (Northeastern Univ., Boston, MA 02115)

The relation between simple loudness adaptation and exposure time was determined for continuous pure tones at eight frequencies from 0.125 to 16 kHz and five sensation levels (SL) from 5 to 40 dB. Adaptation was measured over a 6-min exposure period in eight young listeners with normal hearing by the method of successive magnitude estimation. Listeners judged loudness every 20 s by assigning a number that matched the loudness of the tone. At all frequencies, loudness declined over time, more so at low sensation levels than at higher ones, and more so at frequencies above 8 kHz than at lower frequencies, especially at 5- and 10-dB SL and at 16 kHz. Most of the adaptation occurred within the first 3 min of exposure, slowly approaching asymptote around 6 min. Additional measurements at 60 dB SL at three frequencies showed that loudness declined after 6 min by 16% at 2 kHz and by as much as 26% at 12 kHz. Results obtained by loudness matching closely agreed with those from magnitude estimation. Quantitative analyses revealed a clear relation between published excitation patterns based on masking patterns [E. Zwicker and H. Fastl, *Psychoacoustics* (1990)] and the current adaptation data. [Work supported by NIH.]

2:45

2pPP8. Deriving the Weber fraction from loudness functions. William S. Hellman (Dept. of Phys., Boston Univ., 590 Commonwealth Ave., Boston, MA 02215) and Rhona P. Hellman (Northeastern Univ., Boston, MA 02115)

In a previous paper [W. S. Hellman and R. P. Hellman, J. Acoust. Soc. Am. **91**, 2380(A) (1992)], Weber fractions for intensity discrimination were derived from their concomitant pure tone loudness functions in normal hearing. The calculational procedure employed a generalized McGill–Goldberg model. This work extends these findings to more frequencies, to broadband noise, to tones masked by high-pass noise, and to forward masking. Weber functions generated by the model are compared to empirical data for the various experimental conditions and stimuli. In all cases, the calculated Weber functions capture the overall shape of the measured intensity-jnd data over a wide stimulus range. Not only does the model produce the near-miss relation for pure tones in quiet, it also predicts Weber's law for broadband noise, a rising characteristic in the Weber function above 60 dB SPL for a tone in high-pass noise, and the midlevel hump observed in forward masking. These results show that there is sufficient information in the loudness function to recover the associated intensity-jnd function. This implies that loudness is the primary decision variable for intensity discrimination.

3:00–3:15 Break

3:15

2pPP9. Reaction time adaptation: Hick's law may bias the outcome. T. Goldman, E. M. Weiler, D. E. Sandman, and J. M. Davis (ML #379, Communication Sciences and Disorders, Univ. of Cincinnati, Cincinnati, OH 45221)

Auditory adaptation is characterized by a decrease in apparent loudness over several minutes of time. This is true for magnitude estimated loudness adaptation measured either ipsilaterally [Weiler *et al.*, Br. J. Audio. **15**, 201–204 (1981)] or induced binaurally [Botte *et al.*, J. Acoust. Soc. Am. **72**, 727–739 (1982)]. The extent of ipsilateral adaptation is also a function of time of day [Sandman *et al.*, J. Aud. Res. **22**, 65–69 (1982)], as well as duration of exposure [Weiler and Cobb, J. Aud. Res. **22**, 233–239 (1982)]. The dB adaptation by the classic Simultaneous Dichotic Loudness Balance procedure progresses over time [Hood, Acta Oto-Laryngol. Suppl. **92**, 1–57 (1950)]. In 1976, Davis and Weiler [Br. J. Audiol. **10**, 102–106] found that simple reaction time (RT) to a constant intensity, increased reliably after 7 min of exposure, as if the intensity had decreased. Goldman *et al.* [J. Aud. Res. **21**, 13–16 (1981)] and Weiler *et al.* [J. Gen. Psychol. **114** (1987; errata, 1988)] confirmed this effect. However, in a modified design [T. Goldman, Ph.D. dissertation, University of Cincinnati (1985)] the reaction time increased significantly only for the second trial after 15 s of exposure. Thereafter RT values did not differ significantly from the baseline. Hick's law is offered as an explanation, as well as differences in procedure between RT adaptation studies.

3:30

2pPP10. Binaural versus ipsilateral loudness adaptation: A simple relationship after all? Robert Tannen, Ernest M. Weiler, Joel S. Warm, William N. Dember, and David Sandman (Mail #379, Psychoacoustics Labs., Univ. of Cincinnati, Cincinnati, OH 45221)

Weiler and Hood [Audiology **16**, 499–506 (1977)] reviewed a model of loudness adaptation proposed by Hood. They found that the model could predict adaptation measured by the traditional simultaneous dichotic loudness balance (SDLB) procedure within a few dB. However, failure to correlate this method with ipsilateral loudness adaptation forestalled attempts to convert the model to monaural/ipsilateral adaptation. The present paper reports success at finding a means of converting the Hood/Weiler model to results from the ipsilateral comparison paradigm. Results from the magnitude estimated ipsilateral procedure appear to be mathematically related in a simple way to adaptation expected from the binaural SDLB procedure. Pertinence of an interpretation based on powers of 2 will be discussed.

3:45

2pPP11. Ipsilateral comparison paradigm loudness adaptation: Is contrast hidden in complexity? Keith Jones, Ernest M. Weiler, Joel S. Warm, William N. Dember, and David E. Sandman (Mail #379, Psychoacoustics Labs, Univ. of Cincinnati, Cincinnati, OH 45221)

Dange *et al.* [J. Gen. Psychol. **120**, 217–244 (1993)] investigated Weiler's ipsilateral comparison paradigm (ICP) and determined that the results could not be explained as a function of simple auditory contrast. This study introduces a mixed referent condition, wherein increasing and decreasing referents alternate in the same trial. Results from all conditions demonstrated that both base and referent tones adapt under both levels of intensity (50 and 70 dB). If contrast were the dominant influence, adaptation would not be expected in mixed or decreasing referent conditions, nor should the referents themselves adapt. Thus contrast would not explain the results. However, certain preliminary *post hoc* analyses may reveal an influence due to contrast. Alternately, the simple referent conditions show correspondence with a model for binaural adaptation [Weiler and Hood, Audiology **16**, 499–506 (1977)]. This analysis shows that the average for simple referent ICP adaptation differs from binaural adaptation only by a power of 2.

4:00

2pPP12. Overview: Modeling ipsilateral loudness adaptation as an indicator of peripheral function. Ernest M. Weiler, Jon Temple, David E. Sandman, Teri Huber (ML #379, Psychoacoustics Lab, Univ. of Cincinnati, Cincinnati, OH 45221), and Maureen Korman (Good Samaritan Hospital, Cincinnati, OH 45221)

Weiler and Hood [Audiology **16**, 499–506 (1977)] and others found that a highly accurate model could be constructed to account for adaptation found by the classic method of simultaneous dichotic loudness balances. However, when the ipsilateral comparison paradigm (ICP) was devised to avoid complications of binaural/interaural effects, the SDLB model did not seem to fit [Weiler *et al.*, J. Gen. Psychol. **114**, 411–421 (1987); errata, 1988] and others. Recently, studies by R. Tannen and K. Jones [unpublished MA theses in progress, University of Cincinnati (1996)] seem to have provided the basis for a conversion from the SDLB model to ICP results. A critical assumption is that binaural loudness effects are double that of ipsilateral adaptation. Work by the authors and others will be reviewed with discussion of the strengths and weaknesses of the model for ipsilateral adaptation.

4:15

2pPP13. A theory of the principal monaural pathway. I. Pitch and time perception. Neil P. McAngus Todd (Dept. of Psych., Univ. of Manchester, Manchester M13 9PL, UK)

A theory of the principal monaural pathway is described according to which temporal information is spatially coded on three dimensions roughly corresponding to the levels of, respectively, the cochlea, the inferior colliculus, and the auditory cortex [C. Schreiner and G. Langner, "Coding of temporal patterns in the central auditory nervous system," *Auditory Function: Neurobiological Bases of Hearing*, edited by G. M. Edelman *et al.* (Wiley, New York, 1988), pp. 337–361]. This theory is expressed in the form of a computational model which simulates peripheral and central processing and which has the following main components: (1) a one-dimensional linear bandpass filter-bank to simulate the cochlea; (2) a two-dimensional amplitude-modulation (AM) bandpass filter-bank to simulate the colliculus; (3) a three-dimensional AM bandpass filter-bank to simulate layer IV receptive cells in the cortex; (4) a cortical cross-channel correlation mechanism; and (5) a central pattern recognition mechanism. Overall, periodicity pitch (i.e., periods of up to about 1000 Hz) is primarily associated with subcortical processing whereas time and rhythm (i.e., periods of up to about 20 Hz) are primarily associated with cortical processing. The model is applied to the following phenomena: virtual pitch shift, musical chords, the psychophysical law for interval discrimination, and the filled interval illusion.

4:30

2pPP14. A theory of the principal monaural pathway. II. Rhythm, streaming, and comodulation masking release. Neil P. McAngus Todd (Dept. of Pscyh., Univ. of Manchester, Manchester M13 9PL, UK)

The model [N. Todd, "A theory of the principal monaural pathway. I. Pitch and time perception," J. Acoust. Soc. Am. (these proceedings)] proposes to account for auditory streaming by a cross-correlation mechanism modeled as an array of cortical columns. A column contains excilatory and inhibitory interneurons and pyramidal cells which receive input from both the thalamus and from neighboring columns. For columns with coherent thalamic inputs, the outputs of the pyramids sum across frequency because local interneurons compute a correlation between the AM transform of the inputs of the local and remote columns, and selectively gate the input to the pyramidal cell. (1) Grouping by frequency proximity—for columns separated by less than a critical band, thalamic inputs are unresolved. (2) Grouping by temporal proximity—at repetition rates below the mean of the AM distribution, AM fundamentals are less well represented; at higher repetition rates the AM harmonics are more separated. (3) Temporal development—where the AM spectra take time to develop. The AM coherence mechanism also gates the inputs to disjoint subpopulations of a low-pass mechanism, and thus also accounts for comodulation masking release, since backward masking will only take place within a single subpopulation.

TUESDAY AFTERNOON, 14 MAY 1996 — REGENCY D, 1:00 TO 5:00 P.M.

Session 2pSC

Speech Communication: Prosody

Ann R. Bradlow, Chair
Psychology Department, Indiana University, Bloomington, Indiana 47405

Contributed Papers

1:00

2pSC1. Timing of fundamental frequency control in word accent of the common Japanese. Hiroya Fujisaki, Sumio Ohno, and Tohru Ueno (Dept. of Appl. Electron., Sci. Univ. of Tokyo)

The accentual patterns in Japanese can be consistently described in terms of rapid rises and falls in pitch, and their timing can be quantitatively specified by the onset and offset of the underlying accent command applied to a model for F_0 contour generation [H. Fujisaki and K. Hirose, J. Acoust. Soc. Jpn. (E) **5**, 233–242 (1984)]. The present paper describes a quantitative study on the timing of accent commands relative to the acoustic-phonetic characteristics of the morae and phones in utterances of the common Japanese. The speech material consists of words of varying length, phonetic constituents, and accent types, embedded in a carrier sentence. The timing of the accent command was estimated from the F_0 contour by analysis-by-synthesis, while the timing of phones and morae was detected by referring to the frequency-time-intensity patterns. It was found that the relative timing of the accent command can be consistently described in reference to the onset of vowel rather than the onset of a mora when the latter starts with a consonant. Systematic variations in the timing due to phonetic constituency and accent type are formulated as rules for speech synthesis.

1:15

2pSC2. Nasal duration and amplitude as a function of stress and prosodic phrasing in Estonian. Matthew K. Gordon (Dept. of Linguist., UCLA, 405 Hilgard Ave., Los Angeles, CA 90024)

Cross-linguistically, segments are often lengthened and/or strengthened at prosodic boundaries, where the degree of lengthening/strengthening is generally correlated with the level of the boundary: The larger the boundary, the longer/more strengthened the segment. Stress often induces similar lengthening/strengthening. In this study, the duration of Estonian nasals was measured in utterance-, word-, and syllable-initial position in the onset of unstressed and stressed (primary and secondary) syllables. Nasal amplitude relative to the immediately following vowel was also measured in these contexts in order to draw inferences about the effect of stress and prosodic boundaries on nasal articulation. Preliminary results revealed that duration and amplitude are a function of both stress and, to a lesser extent, level of the prosodic boundary. Nasal duration was greater as stress level increased. However, nasal amplitude was greater in the onset of unstressed syllables immediately *following* stressed syllables than in the onset of stressed syllables. Regarding phrasal boundaries, word-initial nasals were longer than syllable-initial nasals, but only in unstressed environments, supporting an interaction between stress and phrasing. Furthermore, utterance-initial nasals were longer than syllable-initial nasals, but had less amplitude, suggesting strengthening (increased "consonantality") in utterance-initial position.

1:30

2pSC3. A comparative study of focus realization in three Swedish dialects. Robert Eklund (Telia Res. AB, Spoken Language Processing, S-136 80 Haninge, Sweden)

State-of-the-art speech recognition and speech translation systems do not currently make use of prosodic information. Utterances often have one or more constituents semantically focused by prosodic means and detection of the focus/foci of an utterance is crucial for a correct interpretation of the speech signal. Thus, a semantic model of focus should be linked to a model describing the acoustic-phonetic correlates of the speech. However, variability exists at both the semantic and the prosodic ends. Semantically different kinds of foci might be associated with specific prosodic gestures. Also, a semantically specific type of focus might be realized in different ways in different varieties of a given language since general intonational patterns vary between dialects. In this paper, focus realization in three different dialects of Swedish is investigated. Subjects from Stockholm, Göteborg, and Malmö recorded three sets of four sentences where focus was systematically put on four different constituents by having the subjects answer wh-questions. Since Swedish is a language with two tonal accents, words with these accents both in and out of focus were included. Dialectal as well as individual variation in focus realization is described with emphasis on invariant and optional phenomena.

1:45

2pSC4. Pitch range and focus in Hindi. James D. Harnsberger (Program in Linguist., Univ. of Michigan, 1076 Frieze Bldg., Ann Arbor, *MI 48109) and Jasmeet Judge* (Univ. of Michigan, Ann Arbor, MI 48109)

A study of intonation in Hindi included sentences, read by six native speakers, in which different individual words were emphasized or focused. Hindi resembles some other languages (English, Bengali, Korean) in marking a focused constituent prosodically by reducing the normal prominence given to neighboring constituents. In Bengali and English, all constituents preceding a focused word are deaccented, failing to show typical contours [Hayes and Lahiri, Nat. Lang. Ling. Theory **9**, 47–96 (1991)]. The reverse occurs in Korean, with following constituents being deaccented (Jun 1993, dissertation). In Hindi, deaccenting of following constituents is also observed; however, the phenomenon is characterized by a dramatic compression of the speaker's pitch range following the focus word. For narrow pitch ranges, compression effects on $F0$ realization are equivalent to those of deaccenting. For wider ranges, following constituents show the typical rising contours over content words found in declarative utterances without focus, though at a narrower range than for the focus word. The theoretical significance of Hindi focus is to illustrate language-specific differences in marking focus while supporting a general cross-linguistic strategy of reducing local prominences. These data also provide additional evidence for the head-initial (Hindi, Korean) versus head-final distinction in prosodic phrases (Bengali, English).

2:00

2pSC5. Foreign-accented rhythm and prosody in reiterant speech. Keiichi Tajima (Dept. of Linguist., Cognit. Sci. Program, Indiana Univ., Bloomington, IN 47405), Jonathan Dalby (Commun. Disord. Technol., Inc., Bloomington, IN 47404), and Robert Port (Indiana Univ., Bloomington, IN 47405)

The prosodic characteristics of utterances produced by second-language learners were investigated using reiterant speech (RS), a stylized form of speaking in which every syllable is replaced with a standard syllable such as [ma], so that the phrase "*the table*" is pronounced "*ma MAma.*" Native speakers of Chinese and Spanish produced RS versions of short phrases in English and in their native language. Preliminary phonetic analysis shows that the English RS tokens were less accurately produced than were the native-language tokens. A subset of the English tokens were then used as stimuli in a perception test in which native English listeners heard each RS phrase and judged whether it "matched" or "did not match" an English phrase presented to them visually. The stimulus set consisted of English RS produced by both non-native and native speakers. Preliminary results indicate that listeners were better at judging the match/mismatch of the native English RS tokens than the non-native tokens. This result indicates that there are perceptually relevant differences between native and foreign-accented RS, and suggests that practice in producing more authentic English RS might be a useful method of teaching aspects of English prosody to second-language learners. [Research supported by NIH-NIDCD Grant No. 2R44DC02213-02.]

2:15

2pSC6. Interactive prosody training workstation. George D. Allen (College of Nursing, Michigan State Univ., E. Lansing, MI 48824) and V. Paul Harper (Harper & Assoc., W. Lafayette, IN 47906)

Few devices exist to aid in the training of pitch, intensity, and rhythm of speech. The Interactive prosody training workstation (PW) employs state-of-the-art technology to assist users in achieving clinician- or client-programmable targets in each of these features. Two training interfaces are currently implemented. The simpler displays F0 and/or intensity in real time as a fluctuating one- or two-dimensional display. Smoothing can be adjusted to accommodate varying levels of voice variations. With the more advanced interface, model utterances are presented using stored LPC-coded speech. The user's response is then compared to the model, using any of a wide variety of scoring methods. The user's response may be played, in comparison to the model, as often as desired. A demonstration tape will be played showing one hearing impaired individual with a cochlear implant using the PW first to find his appropriate F0 register and then practicing pitch gestures appropriate for speech. Strong carryover of learning to later sessions is demonstrated. [Work supported in part by an SBIR grant from NIH.]

2:30

2pSC7. Perceiving the difference between spontaneous and read speech: The role of physical duration. Robert E. Remez, Jennifer S. Lipton, and Jennifer M. Fellowes (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027-6598)

Despite the greater average duration of spontaneous sentences relative to matched fluently read sentences, it is difficult to identify the physical basis for determining whether an utterance is spontaneous or read. The present experiment compared judgments of spontaneity with judgments of duration for the same speech samples. The test material comprised 25 sentence pairs in which one sentence was spoken spontaneously and one sentence was read, both produced by the same speaker. On each trial, subjects identified the sentence of the pair that was spoken spontaneously, and the sentence that was longer in duration; tests were blocked by judgment type. The greater the absolute duration difference was between the two sentences, the better subjects performed on the duration judgment. The physical differences in duration did not predict the spontaneity judgment, nor did the duration judgment correlate with the spontaneity judgment. The results suggest that physical duration alone is not an effective attribute underlying the perceptual differentiation of spontaneous and read utterances. [Work supported by NIDCD.]

2:45–3:00 **Break**

3:00

2pSC8. Detection of sentence accents in a speech recognition system. Per Sautermeister and Bertil Lyberg (Telia Res. AB, S-136 80 Haninge, Sweden)

Speech recognition systems do not usually utilize prosodic information, i.e., information signaled by segmental duration and the fundamental frequency contour of the speech signal. The acoustic manifestation of prosody is, more often than not, considered as a disturbance in current statistical approaches to the speech recognition problem. The detection and transformation of sentence accent in, e.g., spoken language translation systems, will enable stress on a certain word in one language to be transformed into a suitable representation of corresponding constituents in the other language and satisfy the same semantic goal. In this study, a system for automatic detection of sentence accents to be used in speech recognition systems, is presented. The fundamental frequency is extracted from the speech signal and an estimated frequency declination is subtracted from the actual fundamental frequency in order to give a normalized representation of the variations. These fundamental frequency variations are given in musical intervals. The interpretations of sentence accents are carried out from this normalized manifestation of the fundamental frequency. Both the system architecture and some preliminary results will be shown.

3:15

2pSC9. Speech-rate effects on the perception of second formant transitions. Alexander L. Francis and Howard C. Nusbaum (Dept. of Psych., Univ. of Chicago, 5848 S. Univ. Ave., Chicago, IL 60637)

Research on context effects in speech-rate normalization generally focuses on the perception of phonetic categories distinguished by temporally defined cues like VOT or F2 transition duration. For example, a syllable that is heard as [ba] in the context of a slowly spoken carrier sentence will likely be identified as [wa] in a quickly spoken context [Minifie *et al.*, J. Acoust. Soc. Am. Suppl. 1 **62**, S79 (1977)]. This suggests that the category boundary between [b] and [w], when based on an F2 transition duration cue, is interpreted relative to the talker's current speaking rate. One implication of prior research is that speaking rate changes only affect the perception of durational cues. However, research on phonetic cue trading suggests that temporal and spectral cues are perceived together in an integrated fashion based on phonetic knowledge. Since changes in speaking rate may restructure spectral as well as temporal cues in production, listeners may show similar context effects for perception of these cues. This study examines the effect of changing speaking rate context on the perception of [b] and [w] stimuli distinguished by variation in transition duration and extent of frequency change. Implications for theories of speech-rate normalization are discussed.

3:30

2pSC10. Vocal expression of emotion is associated with spectral properties of speech. Jo-Anne Bachorowski (Dept. of Psych., Wilson Hall, Vanderbilt Univ., Nashville, TN 37240) and Michael J. Owren (Reed College, Portland, OR 97202)

Most empirical work in vocal expression of emotion has emphasized prosodic components of speech produced by actors. In the present study, spectral properties of speech samples that were produced during naturally occurring emotional states were examined. Twenty subjects viewed 24 slides selected from the International Affective Picture System. Elicited emotional responses ranged from strongly negative to strongly positive. During the presentation of each slide, subjects provided a free-form description of the feelings and thoughts evoked by the picture, preceded by the stock phrase "This test picture. . . ." Subjects concluded each narrative with the prompt "Next test picture." Analyses of the spectral properties of individual phonemes, drawn from the second stock phrase, showed statistically significant interaction effects involving slide valence and subject emotional intensity. Representations in the frequency space formed by $F1$ and F2, as well as the space formed by $F0$ and $F2$, were related to both current emotional state and to the intensity with which subjects typically experience emotional responses. The obtained effects were of sufficient magnitude to be perceptible, and indicate that an emotion experienced during vowel production can affect the same acoustic cues widely held to be used in a listener's phonemic processing. [Work supported by NIMH.]

3:45

2pSC11. An acoustic evaluation of variation in the overlap of consonant and vowel gestures induced by speaking rate change. Kris Tjaden (Dept. of Communicative Disorders, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182) and Gary Weismer (Univ. of Wisconsin, Madison, WI 53705)

There is a growing body of empirical data supporting a gestural-based view of speech production in which adjacent speech gestures temporally and spatially overlap one another. Recent work suggests that certain acoustic measures are sensitive to changes in the overlap of adjacent consonant and vowel gestures [G. Weismer *et al.*, J. Phon. **23**, 111–126 (1995); E. Zsiga, J. Phon. **22**, 121–140 (1994)]. Utilizing a graded speaking rate task, eight speakers produced 160 repetitions of ten target words embedded in a carrier phrase. $F2$ onset frequency, measured at the consonant–vowel boundary of test syllables, was used to index the degree of spectral-temporal overlap of adjacent consonant and vowel gestures. Regression analyses were used to evaluate the extent to which $F2$ onset predicted temporal variability in $F2$ formant trajectories due to speaking rate change. Results suggest that consonant and vowel gestures do not simply temporally slide away from each other with slowed speaking rate. Rather, modifications in the form and magnitude of vowel gestures are required to account for the present acoustic data. An empirically based acoustic model of variation in gesture overlap induced by speaking rate change is offered. Individual speaker differences also are discussed.

4:00

2pSC12. Competing hypotheses concerning the articulation of stress in English. Jonathan Harrington, Sallyanne Palethorpe (Speech Hearing and Language Res. Ctr., Macquarie Univ., Sydney NSW 2109, Australia), Janet Fletcher (Univ. of Melbourne, Victoria, Australia), and Mary E. Beckman (Ohio State Univ., Columbus, OH 43210-1298)

Previous studies have suggested two different hypotheses concerning the articulation of stress contrasts in English. Examination of low vowels surrounded by labial consonants generally shows lower jaw in accented as opposed to unaccented syllables, a result originally interpreted as a lower and hence louder vowel [Edwards *et al.*, J. Acoust. Soc. Am. **89**, 369–382 (1991)]. On the other hand, a study of nonlow back vowels in alveolar contexts showed higher and backer tongue body in accented syllables, suggesting a more peripheral vowel [de Jong, J. Acoust. Soc. Am. **97**, 491–504 (1995)]. This paper reexamines the two hypotheses by looking at both jaw and tongue for both phonemically low and high vowels. Five speakers of Australian English read 20 tokens of two short discourses which placed the name "Babber" or "Beaber" twice each in accented and deaccented positions. Jaw and tongue positions were recorded simultaneously using a magnetometer. On average, the jaw was lower in all accented syllables, although the difference was smaller for the high vowel. At the same time, the tongue body reached significantly higher positions in accented high vowels, and somewhat lower positions in accented low vowels. Thus the two hypotheses can be reconciled by their application to the two different articulators.

4:15

2pSC13. Jaw displacement and F0 in contrastive emphasis. Donna Erickson (Ctr. for Cognit. Sci., Ohio State Univ., Columbus, OH 43210) and Kiyoshi Honda (ATR Human Information Proc. Res. Labs., Soraku-gun, Kyoto 619-02, Japan and Univ. of Wisconsin, Madison, WI 53705-2280)

This study examines the relationship between highest F0 (within the sonorant portion of the syllable) and maximum jaw displacement (lowest vertical position of the mandible in reference to the maxillary occlusal plane) in utterances with contrastive emphasis. Acoustic and articulatory recordings were made using the x-ray microbeam facilities at the University of Wisconsin [J. Westbury and O. Fujimura, J. Acoust. Soc. Am. Suppl. 1 **85**, S98 (1989)]. Three American English subjects produced question–answer sentences like "Is it 599 Pine Street? No, it's 59FIVE Pine Street," reading from a monitor display with a marking on the digit to be emphasized either in initial, middle, final position, or with no emphasis. The data showed positive correlation between jaw opening and F0 for syllables spoken with contrastive emphasis, but no correlation for nonemphasized syllables. One would expect two opposite biomechanical effects of jaw opening on F0: a positive effect by the action of jaw opening muscle on larynx elevation, and a negative effect by passive hyoid retraction due to jaw lowering. These findings suggest that prosodic contrast may be achieved by reorganization of interactive control of jaw opening and F0. Possible acoustical, physiological, and phonological ramifications of these findings are discussed. [Work supported by NSF SBR-951199B and ATR/ITL, Kyoto, Japan.]

4:30

2pSC14. A kinematic analysis of contrastive stress. Lisa Goffman and Anne Smith (Dept. of Audiol. and Speech Sciences, Purdue Univ., West Lafayette, IN 47907)

The present investigation examined the shape and variation of movement patterns associated with specific stress contexts. Movements of the lower lip were recorded using the Optotrak, a noninvasive system in which light emitting diodes are attached to the moving structure. Eight adult subjects produced versions of the utterance "Buy Bobby a puppy" that varied in stress. Initally, this utterance was produced with normal stress patterns. Then, a scenario was presented that elicited a contrastive stressed form (e.g., to stress "Bobby," the subject responded to the utterances "Don't buy Kathy a puppy. Buy Bobby a puppy."). A standard technique in the analysis of kinematic data is to measure absolute values of peak velocity, displacement, and duration. Analyses revealed that, as expected, increased duration and amplitude occurred in stressed contexts. A second set of analysis procedures involved the evaluation of underlying movement patterns through time and amplitude normalization (i.e., collapsing absolute differences in duration and amplitude). Pattern recognition techniques were then applied, revealing that underlying patterning differed as a function of contrastive stress. [Work supported by NIDCD.]

4:45

2pSC15. Interrogative word is the focus of question in the WH question sentences in Mandarin Chinese? Yueh-chin Chang (Dept. of Foreign Lang., Natl. Tsing-hua Univ., Hsin-chu, Taiwan, ROC)

In nontonal languages, intonation is a very important prosodic feature to distinguish sentence meanings. But in tonal languages, such as Mandarin Chinese, declarative and interrogative sentences both have falling intonation. In this case, interrogative sentences differ from the former by having a question particle in the final position of the sentence or a question word. At the prosodic level, what are the prosodic features which distinguish declarative and interrogative sentences, if these two types of sentences have similar intonational contour patterns? Following Lyons (1977) and Li & Thompson (1979), question words are the focus of questions in an interrogative sentence. They should have a phonetic, acoustic realization more important than other words in the sentence. However, in the primary study about the interrogative word "ji" with third tone (how many) in WH-question sentence, "ji" had a short duration, it had not undergone the 3-tone tone sandhi, and was realized as a neutral tone. This study will examine in detail the acoustic phonetic realizations of interrogative words, to find out their prosodic features and to know how listeners react to these features.

TUESDAY AFTERNOON, 14 MAY 1996 CELEBRATION A, 1:00 TO 2:00 P.M.

Session 2pSPa

Signal Processing in Acoustics: Shirtsleeve (Practical) Statistics

Edith L. R. Corliss, Chair
Forest Hills Laboratory, 2955 Albemarle Street, N.W., Washington, DC 20008-2135

Chair's Introduction—1:00

Invited Papers

1:05

2pSPa1. The need for efficient monitoring of data. Edith L. R. Corliss (Forest Hills Lab., 2955 Albemarle St. NW, Washington, DC 20008)

"Shirtsleeve Statistics" is a name that has been applied to simplified methods for inspecting statistical properties of experimental data. Many of these techniques are in the nature of "nonparametric" statistical operations. Because of their simplicity, they are particularly valuable for monitoring the results of an experiment in progress. Also, study of the inherent relations of the variables can make for a more efficient layout of an experimental procedure. One vital result of this study of observations and their uncertainties is the elucidation of the quantity N, the number of *independent variables* involved in an experiment. Merely counting the number of observations does not suffice. Internal correlations can affect the degree of independence of the observations and thus exaggerate their estimated precision. Techniques are available for simple and timely monitoring of measurement consistencies. It is exceedingly important to keep a running statistical surveillance of data. Inherent variations in the phenomena being measured may obscure temporarily a failure in the measuring or data reduction equipment. Much time and data can be lost. (Remember: Dust swept under the rug raises a lump.)

1:25

2pSPa2. A method for analyzing rating scale data. Harry Levitt (Ctr. for Res. in Speech and Hearing Sciences, Graduate School and Univ. Ctr. of the City Univ. of New York, 33 W. 42nd St., New York, NY 10036)

Subjective ratings are often used in the evaluation of hearing aids, loudspeakers, speech processors, and other systems involving sensory stimuli. The analysis of rating data, however, presents problems since the usual parametric assumptions underlying standard statistical techniques (e.g., *t* tests, analysis of variance) are not applicable to subjective ratings. A nonparametric method of analyzing rating data is described using concepts derived from signal detection theory. In order to compare two sets of ratings, a nonparametric estimate of the area under the ROC curve is obtained by first deriving the correlation function relating the two sets of ratings and then summing over half the range of the correlation function. This statistic, the half summed correlation (HSC), is relatively easy to compute and can be used to test, with known statistical power, for various differences between two sets of ratings (mean differences, variance differences, mean plus variance differences). It is also possible to rank sets of ratings using this statistic and to apply various multidimensional techniques designed for ordinal data. A variation of the analysis of proximities using HSC's will also be described. [Work supported by NIDCD.]

1:45

2pSPa3. Simulations for performance evaluations of a family of fluctuation-based processors. Jacob George (Naval Res. Lab., Code 7176, Stennis Space Center, MS 39529)

A family of processors developed at NRL [R. A. Wagstaff, "The WISPR Filter: A method for exploiting fluctuations to achieve improved sonar signal processor performance," (submitted to J. Acoust. Soc. Am.)] have been shown in data analyses to yield gains in signal-to-noise ratios, and detect weak signals buried in noise. These processors take advantage of the notion that signals from submerged sources (especially those which undergo relatively fewer interactions with the surface) are relatively steady though they may be weak, but ambient noise has greater fluctuations. Simulations using synthetic data have been done to verify the validity of the above premises, and to assess the performance of the family of filters collectively called "WISPR" under several environments. The simulations are also compared with analyses of experimental data. The results of the simulations and the experimental analyses will be discussed. [Work supported by ONR.]

TUESDAY AFTERNOON, 14 MAY 1996 CELEBRATION A, 2:10 TO 5:00 P.M.

Session 2pSPb

Signal Processing in Acoustics: Signal Characterization and Time-Space Analysis

P. G. Vaidya, Chair

Mechanical and Mathematical Engineering Department, Washington State University, Pullman, Washington 99164

Chair's Introduction—2:10

Contributed Papers

2:15

2pSPb1. Analysis of induced chaos in Duffing's equation, using Caseygrams. P. G. Vaidya and C. R. Winkel (School of Mech. and Material Eng., Washington State Univ., Pullman, WA 99164)

In speech signals, a newer type of chaos has been observed. It could be termed as an "induced chaos." In order to see if this type of chaos can be stimulated, in a theoretical setting, the Duffing's equation has been looked at. The parameters of this equation have been chosen in such a way that this would result in an output which is normally nonchaotic. However, the driving amplitude has been modulated, and in turn, the phase and the driving frequency. All of these led to "induced chaos." This chaos differs in certain important characteristics from the usual chaos. It was found that the best way to detect such a chaos was to use the differential trans-spectrograms, or the "Caseygrams." These are obtained by calculating the difference between two specific types of trans-spectral coherences, (TSC's) [P. G. Vaidya and M. J. Anderson, J. Acoust. Soc. Am. **89**, 2370–2378 (1991)]. These two TSC's are specifically developed to measure the interaction of the main harmonics with the subharmonics. These results show that as the modulations increased the induced chaos became more and more apparent in terms of the difference between the TSC's, and thus in the Caseygram. A theoretical explanation of these results is included.

2:30

2pSPb2. Acoustic signal analysis using Gabor-type windows. Izidor C. Gertner, Stephan Lucci, and John Antrobus (Depts. of Comput. Sci. and Psych., City College of New York, New York, NY 10031)

Transients in speech and other acoustic signals are usually irregular and very difficult to analyze because of their short and nondeterministic duration. Therefore the traditional Fourier analysis technique does not give satisfying results. The classical Gabor technique uses a Gaussian waveform window. However, it is not numerically stable and it has slow convergence. In order to do time-frequency analysis of acoustic signals, a window transform is proposed that is derived from a Gaussian using the Zak transform. The resulting window possesses good locality properties in both the time and frequency domains. The computational procedure, which based on the FFT algorithm, is robust and fast. It will be shown that transform yields accurate estimates of onset and offset times for periodic and aperiodic signals as well as for vowel centers and the ramp frequencies of formant transitions.

2:45

2pSPb3. Acquisition of FM waveform parameters from vibration of rotational machine elements using FM wavelet template matching. Michael B. Van Dyke, David C. Swanson, and Karl M. Reichard (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16801)

Vibration signals obtained from rotating machinery exhibit frequency modulated components resulting from torque variations due to wear and irregularities in the elements. The characteristic shape of the FM signal and degree of modulation can be useful metrics for the determination of the nature and extent of element degradation. A nonorthogonal FM wavelet template matching technique is used to quantify these parameters. Results are compared with FM waveforms obtained by signal demodulation, demonstrating fairly good accuracy.

3:00

2pSPb4. Computer imaging features for classifying semivowels in speech spectrograms. Ben Pinkowski and Jack Finnegan-Green (Dept. of Comput. Sci., Western Michigan Univ., Kalamazoo, MI 49008)

Speech spectrograms can be analyzed using computer image processing techniques to yield high recognition rates [B. Pinkowski, Pattern Recognition **26**, 1593–1602 (1993)]. In particular, Fourier descriptors (FD's) have proven useful for characterizing the boundary of segmented isolated words containing the English semivowels (/w/, /y/, /l/, /r/). This study examines the appropriateness of FD's combined with 17 other general features for classifying spectrogram images. The other features include eigenvalues and eigenvectors, gray-level variance and covariance, run-length and chain encodings, and segment size, shape, and compactness. Principal components (PC's) are used for feature reduction on a speaker-dependent data set consisting of 80 sounds representing 20 speaker-dependent words containing semivowels. With eight combined features, including four 32-point FD's and four general features obtained from principal component analysis, a 97.5% recognition rate was obtained using a linear discriminant function. This rate was higher than that observed for any group of features considered separately. [Work supported by NIH.]

3:15

2pSPb5. Use of multiresolution analysis to calculate pitch in presence of noise. Salvador Cerdá (Lab. de Acúst., Dept. de Física Aplicada, Facultad de Ciéncias Físicas, c/o Doctor Moliner, Burjasot 46100, Spain) and J. Romero (Lab. de Acúst., Burjasot 46100, Spain)

The aim of this work was to determine the effect of noise presence at time to evaluate the pitch, and the study of possible improvements. Several algorithms (Terhardt, Duifhuis, Cepstrum, simple FFT, and visual determination) were studied. These algorithms were applied to different Spanish vowels with a progressively increased noise level; conclusions about the precision of each method are presented. In general, the noise causes a loss of accuracy because time-domain data are blurred and spurious components appear in the spectrum. At this point the multiresolution analysis (MRA) technique becomes a useful tool to eliminate the effect of elevated noise level. MRA was used to smooth the vowels analyzed and then the algorithms under study were applied another time. With this procedure an accuracy improvement in all methods that work in the frequency domain was observed.

3:30

2pSPb6. Vowel identification from harmonic contours of vowel centers using an automatic algorithm that eliminates windowing error. Octavio Betancourt and John Antrobus (Depts. of Comput. Sci. and Psych., City College of the City Univ. of New York, New York, NY 10031)

Using successive *non*overlapping 22.5-ms windows, 16-kHz sampling, no filtering, an automatic algorithm locates five successive windows of minimum spectral velocity, describes a subwindow equal to some multiple of the natural period of F_0, and maps the subwindow onto the unit circle, the interval 0, 2π. Consequently, the Fourier analysis is performed on a window where the signal is *exactly* periodic. Because the spectrum contains no extraneous numerical sidebands it is precise, and consists only of natural harmonics in the acoustic signal. The first 32-integer multiples of F_0 are sufficient to describe the spectrum. J. D. Miller's [J. Acoust. Soc. Am. **85**, 2114–2134 (1989)] log $F_0^{1/3}$ shift increases recognition of 12 vowels (men, women, and children) in the Hillenbrand *et al.* [J. Acoust. Soc. Am. **97**, 3099–3111 (1995)] data set from 52% to 75% using a Euclidean classifier (EC—with jackknife). Cosine series (12) were used to compare our Betancourt spectrum (EC: 76%) with the Hamming window (EC: 61%). Quadratic discriminant function analysis + log F_0 (79%) adds only 3% to our best EC result. With this spectrum and a log $F_0^{1/3}$ shift, most vowel information is clearly captured by a simple EC.

3:45

2pSPb7. Extraction and analysis of individual wavefield components from borehole space-time arrays. M. P. Ekstrom (Schlumberger Austin Res., 8311 North RR620, Austin, TX 78720) and C. J. Randall (SciComp, Inc., 5806 Mesa Dr., Ste. 250, Austin, TX 78731)

Spectral analysis approaches have been shown to be useful for the purpose of analyzing dispersion characteristics from borehole acoustical measurements [M. P. Ekstrom and C. J. Randall, J. Acoust. Soc. Am. **98**, 2867 (1995)]. In this approach, the receiver wave fields are processed using a hybrid spectral estimator, with the space-time array first being transformed into the space-frequency domain, and a high-resolution estimator based on the matrix pencil used to estimate the wave numbers at each frequency of interest. In the frequency-wave-number domain, the spectral estimates are of sufficient quality to allow the decomposition of the spectral description into its constitutive parts. The frequency-wave-number plane is segmented to capture the loci of poles for an individual mode or headwave, and the spectral components within this segment extracted from the spectral descriptor matrices. This allows an individual mode to be isolated from the multicomponent wave fields, and the subsequent detailed analysis of its slowness and attenuation. Furthermore, the extracted spectral information can be used to reconstruct the space-time description for the individual mode, thereby providing the complete duality of viewing the individual components in both the space-time and frequency-wave-number domains. This procedure will be demonstrated using both model [C. J. Randall, J. Acoust. Soc. Am. **90**, 1620–1631 (1991)] and field data sets.

4:00

2pSPb8. Wideband higher-order spatial processing of active sonar echoes. Roger F. Dwyer (Information Processing Branch, Naval Undersea Warfare Ctr., New London, CT 06320)

When linear sonar arrays are operated well beyond their design frequency, spatial resolution is reduced at low frequencies and grating lobes appear at higher frequencies based on the second-order spectrum. Because the higher-order spectrum provides more flexibility in choosing the frequency response at the output of a beamformer, the deliterious effects of wide bandwidths can be reduced. This will be demonstrated with active sonar waveforms returned from spherical targets.

4:15

2pSPb9. Model reference signal processing for a laser ultrasonics experiment. James V. Candy, Graham H. Thomas, Diane Chinn (Lawrence Livermore Natl. Lab., Univ. of California, P.O. Box 808, L-495, Livermore, CA 94550), and James B. Spicer (Johns Hopkins Univ., Baltimore, MD 21218)

Laser ultrasonics is an exciting optical methodology of nondestructive evaluation offering a means of detecting flaws in materials especially in hostile areas where contact transducers cannot function such as high temperature environments or awkward areas where the laser is easily directed by mirrors for rapid scanning and measurement. Typical measurement techniques utilize laser interferometers to accurately measure surface displacements. In this paper the feasibility of applying model-reference signal processing techniques are investigated that would improve the performance of a moderate cost, Michelson interferometric measurement system. A model-reference approach is developed to solve the signal enhancement problem for a laser ultrasonics application in nondestructive evaluation. In this problem a sophisticated laser thermoelastic propagation model is used to predict the surface displacement of the specimen under test. Once synthesized, this model displacement response is used as the reference signal in an optimal (minimum error variance) signal enhancement scheme. Both fixed and adaptive processors are considered in this application where it is shown that a significant improvement in signal levels can be achieved over the usual methods to enhance noisy data acquired from a Michelson interferometric measurement system and increase its overall sensitivity.

4:30

2pSPb10. Acoustic-optic hybrid imaging experiment. Anthony D. Matthews and Lisa L. Arrieta (Coastal Systems Station, Dahlgren Div., Naval Surface Warfare Ctr., Panama City, FL 32407-7001)

A series of experiments to determine the ability of laser vibrometers to receive and process echoes from the water surface are described. Imaging of the bottom and objects lying on the bottom is attempted. The excitation is accomplished by a submerged acoustic source operating near 10 kHz. The degradation of the image is quantified as a function of wave height. Experiment results are reported. [Work supported by Coastal Systems Station under Internal Research funding.]

4:45

2pSPb11. Planar acoustic holographic reconstruction by using a moving array. H.-S. Kwon, S.-H. Park, Y.-H. Kim (NOVIC, Dept. of Mech. Eng., KAIST, Taejon 305-701, Korea), and B.-S. Ko (Daewoo Motor Co., Inchon, Korea)

Planar acoustic holography has been well acknowledged as a useful tool to predict whole sound fields in space. In practical applications, however, an insufficient number of microphones has trouble measuring whole holograms, therefore limiting the improvement to construct a better picture of sound. This problem conveys the issue to virtually increase the number of microphones so that it can construct a better, realistic sound field. As one of these measuring techniques, the conventional scanning technique requires a reference and holds the array during the measurement of pressures. Instead, the proposed method measures the hologram continuously by a moving array without a reference. Its basic idea is that the pressure signal of a moving microphone is modulated with the carrier frequency which is the frequency of the sound field. Although this is a time signal, it includes the spatial information of pressure distribution; i.e., hologram, due to a constant moving speed. The bandwidth of modulated signal is directly proportional to the Mach number, therefore generally narrow, a hologram can be successfully recovered unless there is a closely located frequency; not closer than the bandwidth. Also this technique can be directly applied to the hologram measurement of a moving source with a fixed array. This method is demonstrated by computer simulations and experiments.

TUESDAY AFTERNOON, 14 MAY 1996 CELEBRATION B, 1:15 TO 4:50 P.M.

Session 2pUW

Underwater Acoustics: Scattering

John R. Preston, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Chair's Introduction—1:15

Contributed Papers

1:20

2pUW1. Forward scattering at 400 Hz from rough seas in the presence of a range-independent underwater bubble layer. Guy V. Norton, Richard S. Keiffer (Naval Res. Lab.-SSC, Stennis Space Center, MS 39529), and Jorge C. Novarini (Planning Systems, Inc., Long Beach, MS 39560)

Of fundamental interest to the underwater acoustics community is the role that bubbles may play in connection with scattering from the rough air/sea interface. In this paper a high fidelity waveguide propagation and sea surface scattering model known as FEPE-CM [G. V. Norton *et al.*, J. Acoust. Soc. Am. **97**, 2173–2180 (1995)] is applied to the problem of forward scattering from a rough air/sea interface in the presence of a depth-dependent (horizontally statified) bubble layer. The numerical experiments considered were designed to explore the role of upward refraction (due to the presence of the bubbles) on forward scattering from the air/sea interface. Results from these numerical experiments will be presented. Preliminary results show an enhancement to the forward surface scattered field. [Work supported by ONR.]

1:35

2pUW2. Acoustic scattering by a partially buried three-dimensional elastic object. Raymond Lim (Coastal Systems Station, Code 130B, Panama City, FL 32407)

A formally exact transition-matrix solution for the spectral scattering response of an elastic object that penetrates a plane-stratified fluid host is formulated. The field scattered from the segment in each layer is expanded in a global outgoing basis centered on that segment and these segment fields are superimposed at the field point. A manageable structure for the transition matrix is maintained by using the boundary conditions to couple these segment fields to the interior field of the object via a single exterior surface field expansion centered on the origin of the object. The standard set of regular spherical eigenfunctions of the Helmholtz equation are used to expand the exterior surface field. Numerical tests for an axisymmetric spheroid indicate this choice yields a viable solution but convergence is better for flattened shapes (oblate) than elongated shapes (prolate). Examples are presented to illustrate environmental effects on the backscatter by a bubble and an elastic spheroid that penetrate a water/sediment interface. [Work supported by ONR.]

1:50

2pUW3. Scattering from a fluid loaded elastic spherical shell in proximity to a rough interface: Numerical results. Judy Smith and Garner C. Bishop (Naval Undersea Warfare Ctr. Div., Newport, RI 02840)

A null field T-matrix formalism is developed and used to calculate plane-wave scattering from a fluid loaded elastic spherical shell in proximity to sound hard, sound soft, fluid–fluid, and fluid–elastic interfaces with periodic surface roughness. For each type of interface, the Helmholtz–Kirchhoff integral representation of the various scattered pressure and displacement fields are constructed; the surface fields are required to satisfy the appropriate boundary conditions and the scattered fields are required to satisfy the extended boundary condition. Spherical basis functions are used to construct a free field T-matrix for the elastic shell and rectangular vector basis functions are used to construct a representation of the free field T-matrix for the rough interface. The free field T-matrices are introduced into the Helmholtz–Kirchhoff equations for the scattered fields and the null field equations for the shell-interface system and an "exact" analytical solution is obtained. Numerical results are obtained that demonstrate the effects of boundary type, elastic parameters, roughness geometry, amplitude, and slope on the scattered pressure field and on the ability to "see" the scatter from the elastic shell.

2:05

2pUW4. Computing the acoustic field scattered from proud, partially buried, or totally buried cylindrical objects. John A. Fawcett (SACLANT Undersea Res. Ctr., 19138 La Spezia, Italy)

A boundary integral equation method which utilizes the Green's function for a 2-half-space acoustic medium and the Fourier–Bessel representation of the acoustic field within a cylindrical object is presented. This formulation allows for the accurate computation of the acoustic field scattered from a cylindrical object (possibly, with interior layering) which can be above, below, or intersect the waveguide interface. The waveguide Green's function and its radial derivative are expressed in terms of wave-number integrals. The asymptotic (large wave-number) behavior of these integrals is then evaluated in terms of Hankel functions and asymptotic reflection and transmission coefficients. The numerical treatment of these terms must be considered carefully; in particular, for the case of the partially buried cylinder. Spectral backscattering curves are computed for cylinders with varying degrees of burial. Also, the results of two-dimensional full field computations are shown.

2:20

2pUW5. Measurement and localization of interface wave reflections from buried objects. Eric Smith, Preston S. Wilson, Fred W. Bacon, Jason F. Manning, John A. Behrens, and Thomas G. Muir (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

It is demonstrated that interface wave reflections from buried objects can be created with an active seismic interface wave sonar, and measured and localized on a seismic array. The sonar reported here was implemented on a natural beach of the Gulf of Mexico. It employs a monostatic source/receiver geometry with a three-element horizontal receiving array, in which each element is a triaxial geophone (velocimeter). Detailed measurements of propagation loss at the site make it possible to compare reflected signal power with the results of predictions computed in first order, pointlike-scatterer perturbation theory. These propagation measurements, together with phase information, verify that the reflections are interface waves, and measured values agree well with predictions, within experimental and theoretical errors, both of which are well constrained and small. Simulated data are used to compare a beam pattern at uniform shading with experimental results, and it is found that nonuniform shading in the experimental data produces closest correspondence. Independently performed ground truthing and reverberation analysis further provide a consistent and thorough picture of the background sound fields from which these echos must be extracted. [Work supported by the Office of Naval Research.]

2:35

2pUW6. Backscattering mechanisms for reverberation from deep ocean features using ARSRP data taken at the Mid-Atlantic Ridge. J. R. Preston (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

In July 1993 SACLANTCEN participated in an experiment for the acoustic reverberation special research program (ARSRP). The primary objective was to take high-resolution measurements to determine the detailed physical processes dominating the low-frequency scattering from rough topographic features and from deep sediment pond areas. A very detailed set of monostatic and bistatic scattering experiments were conducted (using low frequencies from 200–375 Hz) just west of the Mid-Atlantic Ridge near 26° N and 47° W. This paper presents towed array data recorded by SACLANT Centre's R/V ALLIANCE for comparison with FEPE two way parabolic equation model estimates of selected ARSRP geometries. The receivers were horizontal arrays of 128 elements spaced at 0.5, 1, and 2 m. Source/receiver depths were $\approx$130 and 450 m, respectively. Inferences for monostatic bottom scattering mechanisms are discussed based on the FEPE model from Collins to interpret the data. The modeling work isolates different scattering mechanisms to assess their relative contributions to the overall reverberation levels observed in the data.

2:50

2pUW7. Seafloor acoustic backscattering from different geological provinces in the Atlantic Natural Laboratory. Robert J. Greaves and Ralph A. Stephen (Woods Hole Oceanograph. Inst., 360 Woods Hole Rd., Woods Hole, MA 02543)

The characteristics of acoustic signals backscattered from inside-corner oceanic crust and outside-corner crust are investigated using acoustic reverberation data from the 1993 ARSRP Acoustics Cruise. Specifically, the seafloor dip distribution is compared between areas of each crustal type and correlations are sought between true grazing angle and backscattered signal strength. Beamformed and match filtered acoustic data from the monostatic, wideband, LFM experiment at Site A are used to find the scattering strength corresponding to specific areas of the seafloor. Scattering strength determined from intersecting beams of different segments are summed to reduce left–right ambiguity. At the scale of analysis, high scattering strengths are found to correspond to steep flanks of seafloor features and can be used to map their shape and orientation. Some of these features are characteristic of specific crustal regions. Scattering strength is found to increase with increasing effective grazing angle at a rate of about 0.1 ± 0.01 dB/degree. However dip values are so scattered that the seafloor dip, on the scale of a few hundred meters cannot be used to predict backscatter intensity. Some other characteristics of steeply dipping areas, such as subsurface heterogeneities or smaller scale surface features, strongly influence the level of backscattered signals. [Work supported by ONR.]

3:05–3:20 Break

3:20

2pUW8. Vertical spatial and temporal coherence of ocean bottom reverberation. Dajun Tang, George V. Frisk, Cynthia J. Sellers (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Dan Li (MIT/WHOI Joint Program in Oceanogr./Oceanograph. Eng., Woods Hole, MA 02543)

An experiment to study acoustic backscattering from deep-ocean sediments was conducted in July 1993 as part of the Acoustic Reverberation Special Research Program (ARSRP). An acoustic source transmitting chirp signals in the frequency range 250–650 Hz and a 24-element vertical receiving array attached to the source were suspended near the seafloor over a sediment pond in the vicinity of the Mid-Atlantic Ridge. In a previous paper [Tang *et al.*, J. Acoust. Soc. Am. **98** 508–516 (1995)], a study on bottom scattering due to sediment volume inhomogeneities in this area was presented. Here, the spatial and temporal coherence of the scattered field using the data collected on the vertical array will be examined. The scatterers are modeled as point scatterers. It is shown that when the coherently reflected signals due to sediment layering are removed from the measurements, the results of the point scatter coherence model agree satisfactorily with data. [Work supported by ONR.]

3:35

2pUW9. A note on scattering by a stack of rough interfaces. Dajun Tang (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Scattering by a stack of random rough interfaces is studied. Such interfaces are common in both shallow water and deep ocean sediments due to the natural sedimentation processes. When the roughness of each interface is small, first order perturbation theory of Bass and Fuks [*Wave Scattering from Statistically Rough Surfaces* (Pergamon, New York, 1979)] is employed to formulate the scattered field due to such an interface. If the distances between interfaces are small compared to the acoustic wavelength and the variations of sediment properties are small, it is found that the collective scattered field due to a stack of random interfaces can be related to that due to a weak scattering volume where sound speed and density are randomly varying. Thus low-frequency subbottom scattering due to either weak volume inhomogeneities or a stack of rough interfaces can be formulated in a unified fashion. [Work supported by ONR.]

3:50

2pUW10. Bottom backscatter measurements at 100 and 200 kHz with high angular resolution. Anthony F. Parkinson and Stuart D. Anstee (Maritime Operations Div., Defence Science and Technol. Organisation, Wharf 17, Pirrama Rd., Pyrmont, NSW 2009, Australia)

Measurements of the variation of acoustic bottom backscatter with incident angle have been made at 100 and 200 kHz at sites in Sydney Harbour, Australia, with bottom types varying from sand waves to mud. The measurements were made with transducers mounted on a rotating arm with a radius of approximately 2.4 m. Angular resolution of approximately

2p TUE. PM

0.2° was achieved over a range of nearly 90°. Twin cameras attached to the rotating arm allow the recording of stereo images of the bottom in favorable cases, giving a model of the scattering surface allowing a comparison of measured backscatter with that numerically estimated from surface scattering alone.

4:05

2pUW11. Angular variation in the simulated reflection and scattering properties of granular poroelastic ocean sediments. Dennis J. Yelton and Nicholas P. Chotiros (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

Reflection and scattering properties of an inhomogeneous poroelastic medium were studied via numerical simulation. The inhomogeneous medium was modeled as an ensemble average of randomly layered poroelastic material. Each layer represented a granular material of a particular grainsize. The thickness of each layer was related to the associated grainsize and porosity by a conservation of mass relationship. Lateral variations in grainsize were approximated by performing a coherent ensemble average of results from several realizations of the randomly stratified medium. Poroelastic medium parameters were chosen to represent water-saturated sand. The mean and standard deviation of the grainsize distribution were chosen to match existing experimental data in order to test the model. Specifically, the inhomogeneous medium was modeled as bounded by a homogeneous water half-space on the source side, and a homogeneous poroelastic half-space of equivalent average porosity on the other side. Reflected signals were computed for high-frequency plane waves incident at various grazing angles. Coherent and random components of the reflected signal were calculated. The coherent part was related to the reflection coefficient. The random component was related to the scattering strength of the medium. Results were compared with existing experimental data. [Work supported by ONR.]

4:20

2pUW12. High-frequency bistatic scattering from elastic seafloors. Anatoliy N. Ivakin (Andreev Acoust. Inst., Shvernika 4, Moscow 117036, Russia) and Darrell R. Jackson (Univ. of Washington, Seattle, WA 98105)

A high-frequency model of bistatic sound scattering is considered assuming an elastic seabed with spatially fluctuating bulk parameters and a slightly rough surface. The first-order perturbation solution is used to obtain an approximate expression for the bistatic scattering strength, different geometric configurations are considered, and the dependence of bistatic scattering strength upon the incident and scattered grazing angles, as well as the "bistatic," or azimuthal angle, is computed for various seabed types. The parameters of the model include mean values for bulk properties and parameters defining the spectra of fluctuations in bulk properties and seabed relief. The influence of shear elasticity is examined by comparison with the case of a fluid seabed having the same density and compressional wave speed. It is shown that shear effects on both roughness and volume components of scattering are small for sands which consequently can be treated as acoustic fluids for realistic shear velocities (approximately 300 m/s and less). For consolidated sediments and rock, shear effects are dominant, complicated, and very sensitive to bottom parameters. Volume scattering exhibits a strong dependence on correlations between the parameter fluctuations as well as their spectral aspect ratio. [Work supported by ONR.]

4:35

2pUW13. Moderate-frequency scattering from layered sediments over an elastic basement. Anatoliy N. Ivakin (Andreev Acoust. Inst., Shvernika 4, Moscow 117036, Russia)

In previous work a high-frequency model for sound scattering by an irregular elastic seabed with volume inhomogeneities and rough water-seabed interface was examined [A. N. Ivakin and D. R. Jackson, J. Acoust. Soc. Am. **98**, 2989(A) (1995)]. Scattering from fluid layered sediments with different kinds of irregularities was also considered earlier [A. N. Ivakin, J. Acoust. Soc. Am. **95**, 2884(A) (1994)]. But, at many sites, the seabed is of a composite-type consisting of an irregular fluid sediment layer or layers covering an elastic basement. In this case, a fluid scattering model is adequate only for sufficient high frequencies such that absorption renders the influence of the elastic basement negligible. At lower frequencies, sound penetration of the sediment increases, and the effects of scattering from the basement as well as additional channels of scattering from within the sediment due to reflection from the basement become important. These effects are considered and their sensitivity to different model parameters is analyzed. Frequency-angular dependencies of scattering strength are calculated for different types of seabed, and possible applications of predicted interference patterns to remote sensing of the seabed are discussed. [Work supported by ONR.]

Session 3aAA

Architectural Acoustics and Musical Acoustics: Directivity of Musical Instruments I

David J. Prince, Cochair
The Talaske Group, Inc., 137 North Oak Park Avenue, Oak Park, Illinois 60301-1336

Richard H. Talaske, Cochair
The Talaske Group, Inc., 137 North Oak Park Avenue, Oak Park, Illinois 60301-1336

Chair's Introduction—8:00

Invited Papers

8:10

3aAA1. Musical instrument directivity in different playing modes, environments, and amplifications. Daniel W. Martin (7349 Clough Pike, Cincinnati, OH 45244)

Directional effects are reviewed for orchestral trumpet and French horn, for piano, and for organ pipes. The implications for each will be discussed relative to different acoustical environments and instrument amplification systems.

8:30

3aAA2. Modes of vibration and directivity of percussion instruments. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Musical instruments in the percussion family radiate sound from many normal modes of vibration. These modes have distinctly different radiation patterns, resulting in a sound field that varies markedly with frequency and with time. Some of the modes of vibration and the resulting radiation patterns obtained from a representative number of percussion instruments, including bells, various types of drums, and cymbals, are described.

8:50

3aAA3. Directivity of a simplified clarinet. William J. Strong and Scott D. Sommerfeldt (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT 84602)

A simplified clarinet was modeled as a closed–open tube of uniform cross section with an open tonehole lattice near its open end. The tube had a length of 50 cm and a diameter of 1.5 cm. The tonehole lattice consisted of five identical toneholes spaced at 2.8-cm intervals. Each tonehole was 0.8 cm in diameter and had an effective length of 1.3 cm. The open end of the tube and each of the toneholes was treated as a simple source, and a corresponding source strength (volume velocity) was calculated at frequencies of interest. The far-field radiated pressure was calculated at a constant radius in a plane that included the tonehole lattice. The pressure was calculated at angular positions from 0 deg (open end) to 180 deg (closed end) for normal mode frequencies. (Circular symmetry about the tube axis was assumed.) Significant directional patterns were seen for frequencies above 2000 Hz. Experiments were performed on a similar structure to measure the directivity and for comparison with the calculations.

9:10

3aAA4. Directional radiation by wind instruments. P. L. Hoekje (Dept. of Phys., Univ. of N. Iowa, Cedar Falls, IA 50614-0150)

As with most musical instruments, the radiation patterns of wind instruments include directional components that can provide the listener with important timbral cues in a room. The most significant radiation comes from vibrations of the internal air column that are transmitted either through the woodwind tone holes or the opening of the brass instrument bell. For each of these, a cutoff frequency can be defined that marks the boundary between low frequencies that are radiated isotropically and the very directional high frequencies. Thus the listener in a room with reflecting surfaces is presented with several "views" of the instrument, and each view has a different spectral distribution. The vibration of the walls of the brass instrument provides another but much smaller directional component. The amplitude of this signal is of the order of 40 dB smaller than the signal radiated directly from the vibrating air column, and so it will usually be masked. However, the longer decay time of this signal increases the chance of its detection. The radiation patterns of this signal are also much different, as well, being quadrupole or higher in order. [Supported by the Iowa Science Foundation.]

9:30

3aAA5. Sound radiation from boxes with tone holes. Gabriel Weinreich (Dept. of Phys., Univ. of Michigan, Ann Arbor, MI 48109-1120)

The normal modes of a hollow box with one or more sound holes, such as forms the radiating element of a typical string instrument, are combinations of "wood modes" and "air modes." Wood modes have an average frequency spacing independent of frequency, whereas the average spacing of air modes is inversely proportional to frequency for frequencies low enough to make the "thin" dimension of the box smaller than half a wavelength, becoming inversely proportional to the *square* of the frequency for frequencies above that. As a result, the density of modes of, say, a violin, is generally dominated by wood modes at low frequencies and by air modes at high frequencies. The interplay of the two types of modes has radical consequences for the directivity of string instruments, in that the directional pattern can change drastically within very small musical intervals, perhaps accounting for the special "flashing brilliance" that such instruments exhibit. This talk will outline the theory of these effects and support it with experimental observations. [Work supported by NSF.]

9:50

3aAA6. The effect of soundpost removal on normal mode radiation efficiency and directivity of a violin. George Bissinger (Phys. Dept., E. Carolina Univ., Greenville, NC 27858)

The radiative properties of individual mechanical normal modes of vibration of a violin can be calculated from experimental modal analysis results using boundary element radiation programs that integrate the Helmholtz equation over all surface elements. This technique, applied to modal analysis results for a violin with and without a soundpost, helps elucidate the acoustic effects of removal of the soundpost both in terms of the radiation efficiency and directivity of the resultant radiation from each mode and the overall acoustic response. Consonant with the experience of players and luthiers, soundpost removal has a major effect on the predicted acoustic response of the violin. Some violin corpus (sans tailpiece and neck-fingerboard) modes that radiate strongly with the soundpost in place show a substantial change in acoustic response when the soundpost is removed, which is dependent either on the strength of their mechanical excitation and/or their radiation efficiency [G. Bissinger, J. Acoust. Soc. Am. **97**, 3154–3164 (1995)]. Particular attention is paid to the predicted acoustic response changes of the first corpus bending modes at 500 Hz.

10:10

3aAA7. Wrapping a violin with planar near-field acoustic holography. Lily M. Wang and Courtney B. Burroughs (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Planar near-field acoustic holography gives one the ability to reconstruct the sound pressure radiated from a source in a three-dimensional field from a single two-dimensional near-field measurement. Traditionally, only one measurement plane has been used which gives the sound radiation in a half-space. In this paper, the near-field acoustic holography is modified to measurements taken on six planes configured as a "box" surrounding the source, thereby giving the sound radiation in all directions. Any source may then be put into this "box"; in this case, a violin is used as the source from which sound radiation patterns can be determined. Preliminary simulations using the modified near-field acoustic holography with six planes are presented. Also, methods of violin excitation with a bowing machine and the positioning of the violin within a measurement frame are described. It is expected that many violins may be tested with this setup, therefore providing a tool for assessing the effects of different modifications to violins on sound radiation. [Work supported by NSF and an AT&T GRPW Grant.]

10:30

3aAA8. Reproduction of directivity pattern using multiloudspeaker source. Philippe Dirogis, Olivier Warusfel, and Reni Caussi (IRCAM, 1 place Igor-Stravinsky, 75004 Paris, France)

A sound source can generally be characterized by three physical properties: timbre, loudness, and directivity. Although loudspeakers give a faithful reproduction of timbre and loudness, their own directivity can conflict with the directivity of the source they aim at reproducing. A general method to optimize the reproduction of any given directivity pattern with a set of independently driven sources is described. At each frequency the synthesized directivity is obtained by looking for the optimal linear combination of the elementary sources, from which the resulting acoustic field can be predicted. This method is applied to a particular source with 12 loudspeakers grouped into a few subsets in order to fit the first spherical harmonics. The comparison between the prediction and the experimental measurements shows excellent agreement. However, the synthesized figures only fit the spherical harmonics in a limited frequency range, which depends strongly on the characteristics of the electroacoustical set (number, position, and directivity of sources). The interest and the limitations of this method are discussed for various types of applications, such as the reproduction of the directivity of instruments, loudspeaker clusters design, and echo cancellation. Finally, an attempt is made to illustrate the perceptual effects of the directivity by analyzing the variations of room acoustics criteria measured with the different synthesized directivity patterns.

10:50–11:05 Break

11:05

3aAA9. Sound intensity patterns from two bass handbells. Edward R. Mansell and Thomas D. Rossing (Dept. of Phys., Northern Illinois Univ., De Kalb, IL 60115)

The sound intensity patterns of two G1 handbells were investigated. One bell is aluminum and the other is traditional bronze. The results indicate that the aluminum bell has a better radiation efficiency for the (2,0) mode fundamental (f−49 Hz). Sound intensity was mapped for the two tuned modes of vibration: the (2,0) and (3,0) modes. The parametric radiation (twice the mode frequency) was significant for both modes, especially the (3,0). The measurement of both active and reactive sound intensity reveals a better picture of the sound field than sound pressure alone. Both of the bells are poor radiators of the fundamental because of their small dimensions compared to the sound wavelength in air. [Both bells on loan from Malmark, Inc.]

11:20

3aAA10. Directivity analysis of piano sounds. Hidetoshi Nakashima (Dept. of Information Eng., Kogakuin Univ., Rm. 5-603, Nakano-cho 2665-1, Hachioji City, Tokyo, 192 Japan), Franck Giron (Ruhr Univ., Bochum D-44780, Germany), and Mikio Tohyama (Kogakuin Univ., Hachioji City, Tokyo, 192 Japan)

The authors investigate the directivity of sounds radiated by a piano and derive spatial impulse responses for representation of the piano directivity in spatial directions. To obtain the radiation patterns of piano sounds, the radiation field was measured in an anechoic room where a microphone array was placed on a spherical surface with a 2-m radius. The microphones used to measure the radiation field were placed every 10 deg from the top of the piano to 130 deg on the lower side of the piano. The microphones were rounded every 10 deg from 0 deg to 350 deg on the horizontal plane. A reference microphone was placed at the center of the sphere on whose surface the measurement microphone array was placed. The spatial impulse responses were obtained as follows: (1) The power spectra of both the radiated sound in all directions and the reference sound at the reference microphone were calculated; (2) the ratio of the radiated spectrum to the reference spectrum was taken; and (3) minimum phase directivity filters were made using the power spectrum ratios obtained through the above procedures. These directivity impulse responses and filters are useful for binaural room simulations including piano sounds, and for synthesizing piano tones in a computer.

11:35

3aAA11. Modes of vibration and directivity of guitars. Eric T. Watson (Watson Acoustical Consulting, 13114 Creekview Park Dr., Houston, TX 77082) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

An acoustic guitar radiates sound primarily from the top plate, the back plate, and through the sound hole, leading to a rather interesting sound field [E. T. Watson and T. D. Rossing, J. Acoust. Soc. Am. Suppl. 1 **83**, S119 (1988)]. The modes of vibration of the guitar body, including the top plate, the back plate, and the enclosed air, and the radiation patterns that result from exciting these modes in an anechoic room, are described.

WEDNESDAY MORNING, 15 MAY 1996 NATIONAL PARKS, 8:00 A.M. TO 12:05 P.M.

Session 3aEA

Engineering Acoustics and Signal Processing in Acoustics: Acoustics for Hands Free Communications

James E. West, Chair

Bell Laboratories, Lucent Technologies, 600 Mountain Avenue, Murray Hill, New Jersey 07974

Chair's Introduction—8:00

Invited Papers

8:05

3aEA1. A DSP implementation of source location using microphone arrays. Daniel V. Rabinkin, Richard J. Ranomeron, Art Dahl, Joe French, James L. Flanagan (Ctr. for Comput. Aids for Industrial Productivity, Rutgers Univ., P.O. Box 1390, Piscataway, NJ 08855-1390), and Michael H. Bianchi (Bell Commun. Res., Morristown, NJ 07960-6438)

The design, implementation, and performance of a low-cost, real-time DSP system for source location is discussed. The system consists of an eight-element electret microphone array connected to a Signalogic DSP daughterboard hosted by a PC. The system determines the location of a speaker in the audience in an irregularly shaped auditorium. The auditorium presents a nonideal acoustical environment; some of the walls are acoustically treated but there still exist significant reverberation and a large amount of low-frequency noise from fans in the ceiling. The source location algorithm is implemented in a two-step process: The first step determines time delay of arrival (TDOA) for select microphone pairs. A modified version of the cross-power spectrum phase method [M. Omologo and P. Svaizer, *Proceedings of IEEE ICASSP 1994* (IEEE, New York, 1994), pp. II273–II276] is used to compute TDOAs and is implemented on the DSP daughterboard. The second step uses the computed TDOAs in a least-mean-square gradient descent search algorithm implemented on the PC to compute a location estimate. [Work supported by a contract with Bell Communications Research.]

8:35

3aEA2. Acoustic echo cancellation for duplex audio communication. Steven L. Gay (Bell Labs., Lucent Technologies, 600 Mountain Ave., Murray Hill, NJ 07974)

This paper reviews the problems associated with the application of acoustic echo cancellation to duplex hands-free audio communication. The more popular approaches to the solution of these problems are reviewed as well. In particular, the problems discussed include the effects of communication channels on the human perception of echo, characteristics of the acoustic echo path, double-talk, and stability. The methods which address these obstacles are fast-converging, low-complexity long adaptive filters, double-talk detectors, and suppression and clipping techniques.

9:05

3aEA3. Low cost methods for mitigating hands-free acoustics problems. Peter L. Chu (PictureTel Corp., 222 Rosewood Dr., Danvers, MA 01923)

Key problems in hands-free acoustics for teleconferencing are acoustic echo cancellation, reverberation, noise, varying amplitude, and restricted audio bandwidth. None of these problems can be completely solved via low-cost methods, but their nuisance can be significantly reduced. For acoustic echo cancellation, a sub-band approach has been found to work well, where the speech is divided up into many bands and echo cancellation occurs individually on each band. Reverberation in speech pickup may be significantly reduced via a simple two dipole, automatically steered array which forms a dipole pattern which may be electronically rotated a full 360 deg. Background noise from fans may be suppressed using spectral subtraction techniques. Varying amplitudes of speech may be addressed by means of a voice-activated AGC (automatic gain control), where the voice detector eliminates pumping artifacts usually found in agressive AGC schemes. Finally the use of 7-kHz digital audio compression schemes alleviates the problems of muffled sounding speech due to the 3.4-kHz restricted audio bandwidth found on telephones.

Contributed Papers

9:35

3aEA4. Talker direction finder system implemented with cardioid beams and DSP algorithm. John C. Baumbauer, Jr., Alan D. Michel, and Jeffrey P. McAteer (Bell Labs., Lucent Technologies, 6612 E. 75th St., Indianapolis, IN 46250)

A prototype talker direction finder (TDF) algorithm has been developed that will rapidly locate a new talker (within 200–400 ms) or track a moving one. For most room configurations, the algorithm has an inaccuracy mean of less than 10° of azimuthal angle and a standard deviation of 7°. The TDF provides full room, 360° coverage. A prototype where the TDF physically rotates a camera toward the talker has demonstrated the TDF's robustness in difficult (i.e., realistic) noisy and reverberant environments. The physical inputs to the TDF are three small and inexpensive omnidirectional microphone elements that are all placed within a 1-in. circle, allowing inconspicuous design into a consumer product. Three μ-law codecs provide Inputs to the digital signal processor where the TDF's new least-mean-squares-type adaptive algorithm resides. The overall system design is summarized, including preprocessing to reduce the effects of noise and reverberation. The adaptive algorithm, which employs redundant "pattern matching" to six cardioid acoustic pickup beams formed from the three microphone element inputs, is then presented. The TDF can provide enhanced solutions for teleconferencing and speakerphone acoustic systems. The algorithm could be used to provide directional information to steer an "adaptive beamformer" for advanced audio pickup applications.

9:50–10:05 **Break**

10:05

3aEA5. A flexible experimental microphone array. James G. Ryan (Inst. for Microstruct. Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

Sound pickup for hands-free communications often requires the microphone to be located at a large distance from the talker. However, in typical environments, the resulting speech signal is corrupted by reverberation and noise which degrade perceived voice quality and impair the operation of automatic speech recognizers. Microphone arrays have been shown to provide substantial improvements in predicted voice quality in computer simulations. To measure microphone array performance in more realistic conditions, a flexible, experimental microphone array has been developed based on digital signal processing technology. This array is flexible enough to permit the implementation of a wide variety of signal processing strategies and array geometries. In this paper the array design will be described and the performance of this array in the variable-acoustics listening room at the National Research Council will be discussed.

10:20

3aEA6. Image-derived second-order differential microphone on a finite baffle. Gary W. Elko (Bell Labs., Lucent Technologies, 600 Mountain Ave., Murray Hill, NJ 07974)

A single first-order dipole microphone placed close to a reflecting plane results in a unidirectional second-order differential microphone. In practice an infinite reflecting plane is not attainable and it is therefore of interest to examine the effects of using a finite reflecting surface on second-order image-derived microphones. This talk will present results obtained in the investigation of a finite reflecting surface size by using the solution to the wave equation for scattering and diffraction from an oblate spheroid. The particular solution that is investigated is for the case of an infinitesimally thin oblate spheroid, the disk. Results show that the response of a first-order bidirectional sensor over the disk is sensitive to sensor location as well as to disk size relative to the acoustic wavelength. For a centrally located image-derived second-order microphone, the differential sensor response approaches a second-order response at a frequency where the reflecting baffle size is at least one-half the acoustic wavelength. Experimental results for both circular and rectangular baffle geometries will also be shown.

10:35

3aEA7. Practical applications of first- and second-order differential microphones in personal computer (PC) multimedia products. Jeffrey P. McAteer (Bell Labs., Lucent Technologies, 6612 E. 76th St., P. O. Box 1008, Indianapolis, IN 46206)

Directional microphones are used in PC voice and telephony applications to reduce the pickup of noise and reverberation, as well as reduce loudspeaker-to-microphone acoustic coupling. Differential-type directional microphones are popular because of their small size and relatively low cost. This talk focuses on how differential microphone technology has been used in new products and solutions developed by AT&T intelligent acoustic systems for the PC multimedia industry.

10:50

3aEA8. Predictability of acoustical effects of microphone ports in the telecommunications equipment. David M. Yeager (Acoust. Technol. Lab., Radio Products Group, Motorola Corp., 8000 W. Sunrise Blvd., Fort Lauderdale, FL 33322)

Low-cost electret microphones are ubiquitous in telecommunications equipment. They are capable of providing nearly ideal response, while often employed as point receivers. However, uniform response is not always optimal, especially for digital applications or microphone arrays, nor is it easily achievable given mechanical packaging constraints. It is therefore essential for a given design to incorporate frequency-dependent effects of the microphone housing. Frequency shaping is typically done electrically, but acoustical filters can be realized by appropriate selection of series or parallel cavities at the microphone. Damping must be accounted for by estimating thermal-viscous effects of the cavity openings, or by introducing flow-resistive materials for which acoustical parameters are known or can be measured. The frequency effects of microphone housing geometries can be modeled using any one of several techniques, the choice dictated by the desired accuracy, complexity of the physical realization, and available resources. In this paper, two prediction methods will be discussed. First, classical lumped- and distributed-element models will be described along with analyses obtained using MATLAB. Second, some preliminary results of numerical models using boundary or finite element methods will be presented.

11:05

3aEA9. Acoustic echo cancellation by short-time Fourier transform and cross-spectrum processing. Takatoshi Okuno (Dept. of Information Eng., Kogakuin Univ., Rm. 5-601, Nakano-machi 2665-1 Hachioji City, Tokyo, 192 Japan), Hirofumi Nakajima, Mikio Tohyama (Kogaquin Univ., Tokyo, Japan), and Hirofumi Yanagawa (Pioneer Electron Corp., Tokorozawa, Saitama, Japan)

Acoustic echo cancellers have problems in estimating room transfer functions (TFs) for acoustic echo paths when the conditions are noisy in the acoustic space. In this article, the possibility of estimating the room TFs is investigated by taking a time-frame-averaged cross spectrum(CS) between the input signals and error signals, which are composed of echo signals through the echo path and the surrounding noise. First, the short-time Fourier transforms (STFTs) and cross spectrum for both the input and error signals are taken. The TFs can be estimated from the ratio of the CS to the power spectrum of the input signal, since it can be expected that the time-frame-averaged CS components between the input signal and the surrounding noise will be zero. It was reconfirmed by computer simulation that the echo signals can be suppressed as the averaging process goes on, even in noisy conditions. The room TFs could be estimated as long as the S/N ratio was about 10 dB. This STFT-CS method requires more calculations than a time domain method, such as the RLS method; however, it can be implemented using general purpose FFT boards. Adaptation for rapid TF changes is going to be investigated in the near future.

11:20

3aEA10. Simulation and cancellation of echoes in a reverberant room. Vineet Mehta, Ming-Yi Lai, and Charles Thompson (Ctr. for Adv. Computation and Telecommun., Univ. of Massachusetts, Lowell, MA 01854)

The evolution of voice communications technology promises high-fidelity teleconferencing systems without encumbering the user with hand-held devices. The freedom of movement provided by hands-free devices yields dynamic acoustic environments and allows for greater separation between source and receiver. Furthermore, these teleconferencing environments may feature communication over multiple audio channels. The sound quality in such environments is typically degraded due to acoustic reflections from nearby walls, people, and objects. These undesired echoes must be removed if one is to maintain a high sound quality. In this work the problem of acoustic echo cancellation is examined. The technique for acoustic echo cancellation is based on sub-band decomposition. For a reverberant room, the source to receiver impulse response is computed by an image- based method. The influence of scattering from objects with high acoustical contrast is computed by a Padé approximants based method. A model for the impulse response resulting from nonideal images, frequency-dependent wall impedance and acoustic scattering is presented. Results compare single and multiple scattering effects. The variations in scattering amplitude due to scatterer motion are examined. The impact of these features on echo-canceller performance is evaluated.

11:35

3aEA11. Considerations in applying noise cancellation techniques to telephones. Alan S. Nasar, Chaim Birman (Dept. of Mech. Eng., Cooper Union, 51 Astor Pl., New York, NY 10003), and Daniel R. Raichel (Cooper Union, New York, NY 10003 and City Univ. of New York, New York, NY)

The intelligibility of voice transmission through a telephone can be severely affected by the presence of excessive background noise. Accordingly, a system was designed to incorporate an auxiliary microphone into the receiver of an ordinary telephone handset. This microphone is strategically placed to pick up ambient noise. The signal from this microphone is subtracted from that of the receiver through a circuit incorporating three op-amps to produce a signal that contains significantly less background noise than the original signal. The corrected signal is then relayed through the telephone in the same manner as that for the receiver in an ordinary telephone.

11:50

3aEA12. Objective and perceptual characterization of microphone arrays for virtual acoustics applications. Olivier Warusfel, Rozenn Nicol (IRCAM, 1 Pl. Stravinsky, 75004 Paris, France), and Yannick Mahieux (CNET, 22307 Lannion Cedex, France)

Microphone arrays represent an attractive sound pickup solution for virtual acoustics applications where a high rejection of the acoustic environment (ambient noise and reverberation) is needed for minimizing artifacts during the postprocessing of the sound (filtering, spatialization, artificial reverberation). This study intends to evaluate, on the basis of simulations and experiments, different prototypes of microphone arrays designed at CNET. The different steps of the array processing (i.e., band filtering of subarrays, sensors weighting, beamforming, and steering) are implemented on a signal processing prototyping environment. This program is used to synthesize the impulse response of the microphone array in a real acoustic field. The input signals are the impulse responses successively measured between a source and each sensor location of the array in real rooms. The objective characterization of the array is based on classical criteria used in room acoustics and on the analysis of the time–frequency envelope of the late reverberation. The perceptual analysis is performed by convolution of the array responses with anechoic signals. A parametrical study is undertaken in order to study the influence of the geometrical and signal processing characteristics of the array. Theoretical and experimental results are compared and discussed. [Work supported by CNET.]

3a WED. AM

Session 3aED

Education in Acoustics: Undergraduate Research Poster Session

Victor W. Sparrow, Chair
Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, University Park, Pennsylvania 16802

Contributed Papers

All posters will be on display from 10:00 a.m. to 3:00 p.m. Contributors will be at their posters from 10:00 a.m. to 1:00 p.m. To allow for extended viewing time, posters will remain on display until 3:00 p.m.

3aED1. Sound velocity measurements in mixtures of air bubbles in castor oil. Colin P. Day and Murray S. Korman (Dept. of Phys., United States Naval Acad., Annapolis, MD 21402)

Experimental measurements of the phase speed are performed in a medium of air bubbles in castor oil. Castor oil has been chosen as the host liquid due to both its density and sound speed being closely matched to that of water. The viscosity of castor oil is about 1000 times greater than water which allows bubbles to rise relatively slowly in a cylindrical container. The phase speed will be measured in an apparatus similar to the standing wave tube setup reported by E. Silberman [J. Acoust. Soc. Am. **29**, 925–933 (1957)]. We are interested in comparing our results with the phase speed theory that was reported by K. W. Commander and A. Prosperetti [J. Acoust. Soc. Am. **85**, 732–746 (1989)]. The volume void fraction β (of air volume to total mixture volume) is estimated by electronic measurements which involve measuring the capacitance of a cylindrical coaxial capacitor in air, in oil, and in bubbly oil.

3aED2. Frequency and amplitude of vibration of reeds from American reed organs as a function of pressure. Philip D. Koopman, Chad D. Hanzelka, and James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

The frequency and amplitude of vibration of a representative sample of reeds from American reed organs have been studied as functions of the (negative) static pressure difference between the interior and exterior of the windchest. Over a pressure range that includes the normal playing pressure of the instrument, the frequency tends to decrease in an approximately linear fashion with increasing pressure difference, but some anomalous effects are observed at higher pressures. Effects of constrictions to the airflow, simulating the effect of a partially opened pallet, have also been studied.

3aED3. A transmission-line model of the vocal tract during fricative production. Jesse Ibarra (Dept. of Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095)

The purpose of this study is to examine the articulatory-to-acoustic mapping of the fricatives /s,sh,z,f,v,th/. The study utilizes MRI-derived vocal-tract area functions for two subjects [Narayanan *et al.*, J. Acoust. Soc. Am. **98**, 1325–1347 (1995)] and an articulatory synthesizer that is based on the analog circuit simulator HSPICE. The synthesizer [Rael *et al.*, J. Acoust. Soc. Am. **97**, 3430(A) (1995)] uses a transmission-line analog model of the vocal tract with the specification of the type and location of dependent or independent sources (voltage or current). Modeling involved placing noise sources in the vicinity of the maximum constriction and at other locations in the vocal tract. A voicing source (KLGLOTT88) was used for the voiced fricatives. Synthesized spectra matched naturally spoken spectra of /s,sh,v,z/ by placing two or three voltage sources near the constriction and near the lips for /s,z,v/, and at 1.2 cm away from the lips for /sh/. The current (monopole) source was not useful in synthesizing these fricatives. Results for /v,z/ were satisfactory using both voicing and noise sources even though /v/ was spectrally flat above 1.5 kHz. The model did not successfully synthesize /f,th/; this may be attributed to the seemingly more complex radiative and turbulent behavior for these sounds. [Work supported by NSF.]

3aED4. Undergraduate research projects in noise and vibration control courses. L. Mongeau and J. D. Jones (School of Mech. Eng., Purdue Univ., 1077 Herrick Labs., West Lafayette, IN 47907)

The integration of projects in undergraduate noise and vibration control courses allows the students to creatively apply the course material to practical problems. The projects also provide the opportunity to investigate some technologies and concepts that are beyond the scope of the lecture material. For a number of years, interdisciplinary projects have been realized in the context of the noise and vibration control courses offered by the School of Mechanical Engineering at Purdue University. Juniors and seniors in mechanical, electrical, and industrial engineering work in teams of three to five students. The objectives of each team are defined in collaboration with selected faculty members and practicing engineers from the local industry, designated as "customers." The projects culminate during the last week of class with oral presentations from each team. In this poster session, the results of selected student research projects realized during the last few years will be presented.

Session 3aNS

Noise: Current Issues in Automotive Noise

Paul R. Donavan, Chair
Noise and Vibration Laboratory, General Motors, GM Proving Ground, Building 24, Milford, Michigan 48380

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. Modern noise and vibration design of internal combustion engines. Michael F. Albright (Roush Anatrol, Inc., Livonia, MI 48150)

In recent years, new engine design for the transportation industry has occurred in an atmosphere of strict compliance with aggressive noise and vibration performance targets. The inevitable result is a growing arsenal of tools, methodologies, and experience in the design of quiet, smooth, high-performance engines. In this discussion, some of the key issues facing engine designers responsible for noise and vibration performance will be introduced. The discussion will focus on the application of some of the newest experimental and analytical tools which are applied to such complex issues as combustion noise, crankshaft to block interaction, valvetrain, and piston to cylinder contact forces. A particular emphasis will be placed on recent developments in combustion noise control and the future of the wholly analytical engine noise and vibration signature simulation model.

8:25

3aNS2. A survey of automobile wind noise. Albert R. George (Sibley School of Mech. and Aerospace Eng., 106 Rhodes Hall, Cornell Univ., Ithaca, NY 14853) and John R. Callister (Cornell Univ., Ithaca, NY 24853)

The study of automobile aerodynamic noise is a field that has grown rapidly in the past decade. Automobile manufacturers and independent researchers now have a better understanding of the mechanisms of automobile aerodynamic noise. Methods of making wind noise measurements have been improved. Some fundamental wind noise reduction measures have been developed. In this paper, the basic mechanisms of automobile aerodynamic noise generation are explained. Current wind noise measurement instrumentation and techniques are reviewed. Some wind noise reduction measures found on today's automobiles are discussed. Finally, a prediction method for an idealized case of automobile aerodynamic noise is shown. Knowledge of these advances will help automobile manufacturers to design quieter vehicles.

8:45

3aNS3. Flow noise in automotive heating, ventilation, and air conditioning systems. Gerald C. Lauchle (Graduate Program in Acoust. and Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

Acoustic noise generated by a typical automotive heating, ventilation, and air conditioning (HVAC) system can be dominated by unsteady flow and turbulence. A comprehensive flow field and acoustic field study has been conducted experimentally on a stand-alone HVAC system operating in its maximum air conditioning mode. Acoustic intensity field mapping was conducted in an anechoic environment. Detailed flow visualizations were conducted underwater on an optically clear system operating at prototypical values of the Reynolds number. The blower is the dominant low-frequency source of noise, while at higher frequencies, additional flow noise sources exist. These include high shear regions within the ducting, separated flows off of flow obstructions, and the exit flow from register vents. Flow visualizations of the blower internal flow field were also conducted on a three-times-larger optically clear model of the blower using underwater particle tracking techniques. The results identify significant regions of inefficient flow, such as blade passage flow separation, reentrant flow, and rotating stall. Collectively, these lead to poor aerodynamic performance as well as noise. [Work supported by Ford Motor Co. and is drawn from the thesis research of former students Timothy Brungart, George Denger, Lori Perry, and Mike Sullivan.]

9:05

3aNS4. Vehicle interior noise. I. Noise paths: Structure-borne and airborne. Alan V. Parrett (Noise & Vib. Ctr., General Motors Proving Ground, Milford, MI 48380) and Richard G. DeJong (Eng. Dept., Calvin College, Grand Rapids, MI 49546)

This paper gives an overview of vehicle interior noise, its characteristics, and control methods. There is a wide variety of noise problems encountered in automobiles due to the sources involved and the range of operating conditions. Depending on these operating conditions, either airborne or structure-borne noise can be dominant. The complexities of automotive structures mean that a variety of analysis techniques needs to be applied to the problems, from the early design stage through to vehicle hardware development. These include both analytical and empirical methods. Examples are given of automotive airborne and structure-borne noise problems in a source, path, receiver framework. Source characteristics (tonal or broadband), problem frequencies, available analysis techniques, and practical solution methods are discussed. Solutions include quieting the source, using isolation at mounting points, and damping and trim materials in the vehicle interior. In practical terms, noise control solutions need to be balanced to comply with overall vehicle cost, mass, and other performance constraints.

9:25

3aNS5. Vehicle interior noise. II. Modeling. Richard G. DeJong (Eng. Dept., Calvin College, Grand Rapids, MI 49546) and Alan V. Parrett (General Motors Proving Ground, Milford, MI 48380)

An overview is presented of several methods available to determine the interior noise in a vehicle under given operating conditions. Issues covered include: how and when these methods can best be applied; the limitations imposed by the methods themselves; and some of the difficulties associated with modeling vehicle structures. Current modeling methods used in automotive noise problems include finite element, boundary element, statistical energy, and power flow methods. Each method has domains of applicability, but more specific to the automotive design process, they have different applicability at different stages in the design cycle. There are practical limitations, such as the lack of detail early in the design process which presents difficulties for deterministic methods at high frequencies. In contrast, later in the process when details are available, there may be the need for more specific answers to design trade-offs than statistical methods can give. Examples are given of some practical problems encountered in vehicle noise modeling, illustrating areas where improvements are still needed.

9:45

3aNS6. On the development and application of automotive sound package materials. Pranab Saha and John Chahine (Kolano and Saha Engineers, Inc., 5095 Williams Lake Rd., Waterford, MI 48329)

It is important to understand not only the noise problem, and the functional and aesthetic issues of an automobile, but also the environmental and competitive issues of the industry to develop appropriate sound packages that will meet the customer requirements. The vehicle manufacturers and their suppliers do so within the bounds of the material and process development technology that is available. The evolution of the basic automobile design and construction also has a significant impact on the sound package development. This may require an entirely new type of material application or an innovative process and product. The paper reviews different types of sound package and process developments that have evolved over the years in some specific applications. These developments not only provide balanced acoustics inside the automobile, but also meet various challenges of the industry.

10:05

3aNS7. Light vehicle exterior noise: Measurement, regulation, tires, and pavement. Paul R. Donavan and Richard F. Schumacher (Noise and Vib. Ctr., General Motors Proving Ground, Milford, MI 48380)

Exterior noise is the only acoustic attribute regulated for passenger cars and light trucks. The primary procedure used to quantify this noise is an outdoor passby test conducted under full-throttle acceleration. Unless specified noise levels are met under this procedure, a vehicle may not be sold in a given market jurisdiction. Recent reduction of European regulatory limits by 3 dB has re-enforced many of the technical challenges faced in designing and testing vehicles to meet these new requirements. These challenges include: better understanding and control of test and environmental variables, more accurate methods of noise prediction, and improved techniques for isolating and reducing individual source contribution. In recent investigations, sound intensity has been employed to isolate tire/pavement interaction noise for vehicles under passby conditions. This has led to the determination that tires can produce significantly higher noise levels under the torque of acceleration than under cruise conditions. As a result, tires are often the major noise source when the total vehicle noise approaches the new regulatory limits. This paper reviews the variables associated with the passby test procedure, the effects of vehicle acceleration on tire/pavement interaction noise, and the needs for improved predictive methods.

10:25–10:35 Break

Contributed Papers

10:35

3aNS8. The effect of surface variability on the short-range propagation of tire/road interaction noise. Denny M. B. Yim (Roush Anatrol, 935 Benecia Ave., Sunnyvale, CA 94086) and J. Stuart Bolton (Purdue Univ., West Lafayette, IN 47907-1077)

The object of this work was to identify the origin of differences between sound levels measured on two track types used for standardized motor vehicle passby noise tests: i.e., the ISO and SAE surfaces. The measured differences could result either from the effect of the road surface on tire/road interaction noise generation, or from surface sound absorption differences. Near-field intensity data measured at the tire/road interface on vehicles in motion were first examined to establish the correlation between surface type and source spectrum and level. Source levels were approximately 1 dB higher at SAE sites than at ISO sites. Two-microphone methods were then used to estimate the acoustic impedance of both the ISO and SAE surfaces. The surface impedances were used together with measured tire source spectra and sound propagation theories to calculate the passby level variability that could be attributed to variations in test site acoustic impedance. It was found that differences in propagation transfer functions for the ISO and SAE surfaces were on the order of only 0.3 dB. Thus it was concluded that the differences in passby levels measured at ISO and SAE sites resulted primarily from differences in tire/road interaction noise generation.

10:50

3aNS9. Two-microphone measurements of the acoustical properties of plane asphalt surfaces in the presence of wind and temperature gradients. Troy J. Hartwig, J. Stuart Bolton (1077 Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-1077), and Denny M. B. Yim (Roush Anatrol, Sunnyvale, CA 94086)

When performing two-microphone measurements of surface properties, an algorithm is used to find the properties that give the best fit between measured and predicted transfer functions between the two microphones. Recently it was found necessary to adjust certain geometrical parameters, e.g., the separation between the source and microphones, to obtain an acceptable fit between measured and predicted data measured over asphalt surfaces. It was hypothesized that it was necessary to allow that variation to account for the effects of wind and temperature gradients. That hypothesis has been confirmed by using ray tracing techniques to simulate the effect of wind and temperature gradients on two-microphone measurements. It has been found that near-surface gradients have significant effects on both the angle of specular reflection and the path length difference between the direct and reflected arrivals. These effects are particularly important when estimating the properties of high impedance surfaces owing to the rapid change of the reflection coefficient at near-grazing angles in that case. Nevertheless, it will be shown that the impedance of asphalt can be estimated accurately if the effect of ray curvature is accounted for when fitting measured and theoretical transfer functions.

11:05

3aNS10. The acoustic characteristics of automotive body seals. Robert Danforth and Luc Mongeau (School of Mech. Eng., Purdue Univ., 1077 Herrick Labs., West Lafayette, IN 47907)

The acoustic characteristics of rubber primary bulb weather seals, used around road vehicle doors to isolate the passenger compartment from water infiltration and air leaks, were investigated experimentally. Short seal samples were mounted within the test section floor of a small, quiet, low-speed wind tunnel adjacent to a soundproof anechoic enclosure. The geometry of the fixture hosting the samples approximated that of a typical vehicle door gap. The results demonstrated the importance of small air leaks and that of the flow-induced Helmholtz resonance of the door cavity on the acoustic performance of the seal assembly. The noise reduction of perfectly sealed systems appeared to be slightly higher with a flow excitation than that measured with a random acoustic excitation using a reverberation room method. The experimental results were compared to the predictions from a two-dimensional lumped-element analytical model based on the assumption that the seal behaves like two limp membranes separated by a closed cavity. The influence of seal compression, rubber properties, and cavity geometry were investigated. [Work supported by Cooper Tire and Rubber Company.]

11:20

3aNS11. A model for predicting roughness of powertrain sound. B. John Feng (Scientific Res. Lab., Ford Motor Co., 20000 Rotunda Ave., Dearborn, MI 48121), Gregory H. Wakefield (Univ. of Michigan, Ann Arbor, MI 48109), and Norman C. Otto (Ford Motor Co., Dearborn, MI 48121)

The semantic dimension of rough/smooth can be an important factor in customer preference of automotive powertrain sound. With this in mind, a new model is proposed for predicting roughness of powertrain sound. The model performs a signal decomposition into critical bandwidth channels, specific roughness prediction for each channel, and a combination of specific roughness across frequency to yield an overall roughness measure. Signal decomposition is performed with a bank of overlapped, critical bandwidth, bandpass filters. Because the signal in each channel is well represented by a small number of harmonically related, narrow-band components, it is assumed that specific roughness can be determined from the peaks in the magnitude spectrum of the envelope signal. A masked threshold estimate from the input signal is used to predict audibility of envelope fluctuation in each filterbank channel. Channels whose envelope fluctuation falls below threshold are assumed not to contribute to overall roughness, and are therefore discarded. Specific roughness values of the remaining channels are combined with a power-law model, yielding an overall roughness measure [E. Terhardt, Acustica **30**, 201–213 (1974)]. Results of subjective tests of powertrain sounds are presented which validate the performance of the model. [Work supported by Ford Motor Company.]

11:35

3aNS12. Modeling of noise spectra of axial flow fans in free field. Shigong Su (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202), Sean F. Wu (Wayne State Univ., Detroit, MI 48202), and Hemant S. Shah (Ford Motor, Dearborn, MI 48120)

A semi-empirical formula for predicting the noise spectra of axial flow fans in a free field is developed. In deriving this formulation it is assumed that sound radiation from an axial flow fan is primarily due to fluctuating forces exerted on the fan blade surface. These fluctuating forces are correlated to the total lift force exerted on the fan blade, and is approximated by pressure pulses that decay both in space and time. The radiated acoustic pressure is then expressed in terms of superposition of contributions from these pressure pulses, and the corresponding line spectrum is obtained by taking a Fourier series expansion. To simulate the broadband sounds, a normal distribution-like shape function is designed which divides the frequency into consecutive bands centered at the blade passage frequency and its harmonics. The amplitude of this shape function at each center frequency is unity but decays exponentially. The decay rate decreases with an increase in the number of bands. Thus, at high frequencies, the narrow bands merge to form a broad band-like spectrum. The noise spectra thus obtained are compared with measured ones from four different types of axial flow fans running under various conditions.

11:50

3aNS13. Sound radiation from a centrifugal blower in a free field. Shigong Su (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202), Sean F. Wu (Wayne State Univ., Detroit, MI 48202), and Morris Y. Hsi (Ford Motor Co., Dearborn, MI 48120)

A new formulation for predicting sound radiation from a centrifugal blower in a free field is derived. The fundamental difference between the present formula and the previously derived ones is that the former considers a new monopole, while the latter assumes the dipole as the primary source. For an axial flow fan or a rotor blade under steady flow condition, the usual monopole source proportional to $(\rho_0 \partial v_n / \partial t)$ is negligible [Siddon, J. Acoust. Soc. Am. **53**, 616–633 (1973)]. However, for a centrifugal blower the monopole source induced by Coliolis acceleration $(\omega \times \mathbf{v}_t)$ cannot be neglected because of a large inclination angle of the impeller. With the inclusion of this monopole source term, the formula yields a dependence of the radiated sound power on the fourth power of the azimuthal speed, which agrees with the experimental data [Maling, J. Acoust. Soc. Am. **35**, 1556–1564 (1963); Neise, J. Sound Vib. **43**, 61–75 (1975)]. The resulting formula can be used to derive the radiated sound power spectrum using the turbulence correlation curve. The spectra thus obtained are compared with the measured ones from a centrifugal blower mounted in a free field. A favorable agreement is obtained in all test cases.

Session 3aPAa

Physical Acoustics and Bioresponse to Vibration and to Ultrasound: Workshop on Therapeutic Applications of Medical Ultrasound III

Francis J. Fry, Chair
414 West Spring Lake Boulevard, Port Charlotte, Florida 33952

This Workshop is an attempt to bring a "small meeting" environment into a big meeting. In addition to the tutorial and special presentations outlined below and in the related two sessions (2aPAa, 2pPAb); there will be significant participation by other workshop attendees; discussion among participants is a major objective of this program. Therefore, a specific timetable will not be followed during the workshop.

Chair's Introduction—8:30

Invited Papers

3aPAa1. Magnetic resonance imaging guided and monitored ultrasound surgery. Kullervo Hynynen (Dept. of Radiol., Brigham and Women's Hospital and Harvard Med. School, 75 Francis St., Boston, MA 02115)

During the past few years there has been an interest in using magnetic resonance imaging (MRI) to guide focused ultrasound beams during noninvasive surgery. MRI offers good soft tissue contrast, allowing the beam to be accurately aimed. Temperature sensitive MRI sequences can be used during low power test pulses to visualize the temperature elevation in the focal zone at temperature levels that do not cause tissue damage. This allows the beam to be aimed, eliminating all potential image or ultrasound propagation distortions. It may even be possible to calibrate the MRI signal to give the temperature elevation and use it as a feedback to modify the power output to assure adequate thermal exposure during every therapy sonication. Finally, MRI allows tissue changes induced by the sonication to be detected. This may be useful after the treatment to establish the boundaries of the thermal damage and is also very useful when experiments are conducted. All of these aspects of MRI guided ultrasound surgery have been investigated using *in vivo* experiments and the results will be reviewed here. The future technical developments and potential clinical applications will be discussed. [Work supported by NIH and General Electric Medical Systems.]

3aPAa2. Renal injury induced by clinical doses of shock waves. Andrew P. Evan (Dept. of Anatomy, Indiana Univ., School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202), Lynn R. Willis, Bret A. Connors (Indiana Univ., Indianapolis, IN 46202), Anne Trout, and James E. Lingeman (Methodist Hospital, Indianapolis, IN 46202)

Both clinical and experimental reports clearly show that shock wave lithotripsy (SWL) causes acute renal effects in a majority, if not all, treated kidneys. SWL-induced acute renal damage results in injury to the nephron, microvasculature, and surrounding interstitium. What has not been established is a set of objectively determined criteria for the safe clinical use of SWL. To accomplish this goal, animal experimentation has been conducted so that the time course and severity of acute alterations could be followed in a model that closely mimics human kidney. We chose the pig to determine the effect (or risk) of the size of the kidney and/or the number of treatments during the same session on the severity of the change in renal hemodynamics. Our data shows the greatest degree of change was induced in the kidney with the least mass and receiving multiple treatments. [Work supported by NIH.]

3aPAa3. Sparse random ultrasound phased arrays for focal surgery. Leon A. Frizzell (Bioacoust. Res. Lab., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801), Stephen A. Goss, Jeffrey T. Kouzmanoff (Labthermics Technologies, Inc., Champaign, IL), and Joseph M. Barich (Univ. of Illinois)

The feasibility of using novel ultrasound phased arrays consisting of array elements larger than one wavelength, minimizing the number of elements in an aperture through a combination of geometric focusing, directive beams, and sparse, random placement of array elements, for tissue ablation applications was examined both theoretically and experimentally. A hexagonally packed array consisting of 108 8-mm-diam circular elements mounted on a spherical shell was modeled theoretically, and a prototype array was constructed, to examine the feasibility of sparse random array configurations for focal surgery. A randomly selected subset of elements of the prototype test array (64 of 108 available channels) was driven at 2.1 MHz with a 64-channel digitally controlled rf drive system. The performance of the prototype array was evaluated by comparing field data obtained from theoretical modeling of the array

configuration to that obtained experimentally via hydrophone scanning. The results of that comparison, along with total acoustic power measurements, indicate that the use of sparse random phased arrays for focal surgery is feasible, and that the nature of array packing is an important determinant to observed performance. [This research was supported by National Cancer Institute Grant No. CA66462.]

3aPAa4. "Popcorn"–and the role of acoustic cavitation in HIFU surgery. Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

3aPAa5. Fetal hemorrhage from lithotripter fields. Diane Dalecki, Sally Z. Child, Carol H. Raeman, Christopher Cox, David P. Penney, and Edwin L. Carstensen (Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Fetal hemorrhages were observed when late term pregnant female mice were exposed to a lithotripter field with peak positive pressures as low as 1 MPa. On the 18th day of gestation, the abdomens of pregnant mice were exposed to 200 pulses from a piezoelectric lithotripter delivered at a rate of $\sim$ 1 Hz. Hemorrhages occurred in the exposed fetuses most often near bony and cartilagenous tissues—head, limbs, and ribs—while soft tissues such as the liver, intestine, and stomach were relatively free of hemorrhage. Thresholds for these effects were $<$ 1 MPa.

3aPAa6. Noninvasive spatially contained high-power ultrasonic system (NSHUS). for blood clot disintegration. Willem Grandia (QMI, Costa Mesa, CA 92627), Robert J. Siegel (Cedars–Sinai Med. Ctr., Los Angeles, CA 90048), and Yoseph Bar-Cohen (California Inst. of Technol., Pasadena, CA 91109-8099)

Ultrasonic waves can be focused from a point outside of the body to an internal blood clot without using invasive techniques. Noninvasive focused ultrasonic techniques for blood clot disintegration may allow for prompt treatment, and eliminate the need for costly and time-consuming invasive catheter-based interventions. Thus this noninvasive approach using therapeutic ultrasound could greatly reduce patient mortality, morbidity, duration of hospitalization, convalescence, and cost of care. An ultrasonic system was designed to operate from 25–200 kHz and provide a localized high power. The profiles of the field and localization effects were analyzed to optimize power containment to the area of the blood clot. To demonstrate system performance tests were initially made to dissolve sugar blocks mounted in an ice/water bath. At this water freezing point, conditions of 0 °C, the rate of sugar dissolution is very slow and it provided a baseline as a measure of ultrasound effect. Various ultrasonic frequencies and power levels were introduced to determine the ultrasonic dissolution/disintegration capability. Simulated blood clots were then exposed to the localized high power and the characteristics of the disintegration as a function of power and frequency were examined. The duration of exposure has been a key parameter in this investigation.

3aPAa7. Therapeutic cardiac ultrasound: Ultrasound accelerated thrombolysis, ultrasound angioplasty, and the "acoustic filter." Richard S. Meltzer (Cardiol. Unit and Ctr. for Biomed. Ultrasound, P.O. Box 679, Univ. of Rochester Med. Ctr., Rochester, NY 14642)

Ultrasound at frequency 1 MHz can accelerate thrombolysis in an *in vitro* system using streptokinase, urokinase, or t-PA. [C. W. Francis *et al.*, J. Clin. Invest. **90** 2063–2068 (1992)]. Ultrasound can also hasten reperfusion due to thrombolytic agents both *in vitro* [D. Harpaz *et al.*, Am. Heart J. **127**, 1211–1219 (1994)] and *in vivo* [R. Kornowski *et al.*, Circulation **89**, 339–344 (1994)]. The mechanism is nonthermal, probably related to cavitation. This type of thrombolysis which is associated with relative low intensity, high-frequency ultrasound is to be distinguished from the high intensity, low-frequency ($\sim$ 20 kHz) ultrasound that has been proposed for "ultrasound angioplasty" [A. Ernst *et al.*, Am. J. Cardiol. **73**, 126–132 (1994)]. Potential clinical applications and problems and limitations to the clinical application of ultrasound accelerated thrombolysis and ultrasound angioplasty will be discussed. An "acoustic filter" has been developed with the goal of removing microbubbles from cardiopulmonary bypass equipment, preventing neurologic damage [K. Q. Schwarz *et al.*, J. Thorac. Cardiovasc. Surg. **104**, 1647–1653 (1992); US Patent No. 5,334,136, 2 August 1994].

3aPAa8. Ultrasound-mediated transdermal delivery of drugs. Samir Mitragotri (Dept. Chem. Eng., MIT, Cambridge, MA 02139)

3aPAa9. Damage to tissue-like structure by wrinkled focused shock waves. Danny D. Howard and Bradford Sturtevant (Graduate Aeronautical Labs., California Inst. of Technol., Pasadena, CA 91125)

Shock waves are focused in extracorporeal shock wave lithotripsy (ESWL) machines to strengths sufficient to fracture kidney stones. Substantial side effects—most of them acute—have resulted from this procedure, including injury to soft tissue. The focusing of shock waves through various layers of tissue is a complex process which stimulates many bio-mechano-chemical responses. This paper presents preliminary results of an *in vitro* study of the initial mechanical stimulus. Planar nitrocellulose membranes of order 10 μm thick were used as models of thin tissue structures. Scattering media were used to simulate the effects of acoustic nonuniformity of tissue and to alter the structure of focusing shock waves. Hollow glass spheres of 65 μm diameter were added to ethylene glycol, glycerin, and castor oil to vary the properties of the scattering media. Multiple layer samples of various types of phantom tissue were tested in degassed castor oil to gage the validity of the scattering media. The scattering media and tissue samples wrinkled the shock structure in a similar fashion. Membranes were damaged by shear stress induced by the altered structure: after about 20 shocks

immersed in the scattering media and after about 100 shocks behind the tissue samples. The mode of failure was in shear with multiple tears about 0.1 cm to about 3 cm depending on the number of shocks and membrane thickness. [Work supported by NIH Grant P01 DK43881-01A3.]

3aPAa10. Application of geometrical shock dynamics to weak shock focusing in extracorporeal shock wave lithotripsy. Joseph E. Cates and B. Sturtevant (Graduate Aeronautical Labs., California Inst. of Technol., Pasadena, CA 91125)

In extracorporeal shock wave lithotripsy (ESWL), weak converging shock waves are focused to shatter kidney stones. Accurate knowledge of the flow field in the focal region is required to understand the source of tissue damage and the primary mechanisms of stone fragmentation. A finite-difference numerical method for geometrical shock dynamics has been developed and used to evaluate the accuracy of this approximate theory for the complex nonlinear processes which occur in shock focusing. The shock dynamics results duplicate the strong, moderate, and weak shock behaviors seen in experiments [B. Sturtevant and V.A. Kulkarny, JFM, 73, 651–671], with good agreement for focal pressures and triple-point path. Similar behavior is observed for the ESWL problem of axisymmetric shock focusing in water. The primary source of error is the fact that shock dynamics always yields Mach reflection at the centerline, so regular reflection in the theory is impossible. Adequate resolution of the focal region is necessary to assess the accuracy of the shock dynamics theory, or of any other theory. The shock dynamics theory and numerical method have also been extended to the more general case of shock propagation into a nonuniform media with free-stream velocity.

3aPa11. Factors involving precision of high-intensity focused ultrasound treatment. Gail ter Haar (Phys. Dept., Inst. of Cancer Res., Sulton, Surrey, U.K.)

WEDNESDAY MORNING, 15 MAY 1996 REGENCY A, 8:25 A.M. TO 12:00 NOON

Session 3aPAb

Physical Acoustics: Bob Beyer Tribute

Murray S. Korman, Cochair
Physics Department, U. S. Naval Academy, Annapolis, Maryland 21402

David T. Blackstock, Cochair
Applied Research Laboratories, University of Texas, P.O. Box 8029, Austin, Texas 78746-7117

Chair's Introduction—8:25

Invited Papers

8:30

3aPAb1. Robert Beyer—A profile. Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520-8286)

From 1942, when he began his doctoral studies, to the present time, Bob Beyer has been a leading voice in the discovery and discernment of acoustical phenomena, the development of educational programs to support students in acoustics, and the dissemination of information regarding acoustics. This introductory presentation will summarize the "parameter space" over which Bob has modestly but impressively held sway. It will include the fields of acoustics to which he contributed, his career as teacher–scholar at Brown University, his activities with the Acoustical Society of America, the American Institute of Physics, and other professional societies and organizations, and his many activities to aid in the internationalization of science, in general, and acoustics, in particular.

8:40

3aPAb2. Acoustics at Brown University: 1956–1958. G. Maidanik (David Taylor Model Basin, NSWC, Bethesda, MD 20084-5000)

In earlier and in those years the physics department at Brown University was mostly acoustics: Profs. B. Lindsay, A. O. Williams, R. Beyer, P. Westervelt, W. Nyborg, R. Morse, and others were all major acousticians in those days, in fact, I converted from low-energy physics to acoustics under their guidance. The news of conversion was drowned, however, by the noise that Sputnik generated in 1957. While the physics and engineering departments were engaged in noise control activities in the populace at large, Prof. Beyer characteristically turned his energies to upgrading high schools' proficiency in physics. In a few months, a "summer teaching for the teachers" was organized at Brown University and a dozen years later we did get to the moon.

8:50

3aPAb3. Robert T. Beyer from his students' perspective. Mark B. Moffett (Naval Undersea Warfare Ctr., New London, CT 06320) and Stephen V. Letcher (Univ. Rhode Island, Kingston, RI 02881)

Professor Robert T. Beyer guided 20 Ph.D. students and 9 Sc.M. students during his career at Brown University. He did a masterful job of preparing his students for successful careers in teaching and research. His genial, outgoing personality ensured good communication with even the introverts among them and his relaxed style (and the close quarters during the Wilson Hall days) encouraged interactions among the students; they often learned the tricks of the ultrasonics trade from each other. He kept a close watch, however, and when he determined that a student was ready to present his first paper at an Acoustical Society meeting (often before the student considered himself ready), he strongly encouraged it and instilled the confidence needed for the task. He personally introduced his students to many of the Society's members, resulting in lifelong friendships and collaborations. Although none of Bob Beyer's students ever succeeded in emulating his ability to tell a story or a joke, they all have benefited greatly from their association with him.

9:05

3aPAb4. Robert T. Beyer's contributions in ultrasonics. Stephen V. Letcher (Univ. Rhode Island, Kingston, RI 02881)

Robert T. Beyer was a major contributor to our understanding of ultrasonic relaxation processes in liquids and solids during the heyday of that field. One could argue that this period ran roughly from the time of his major review article in 1951 [J. J. Markham, R. T. Beyer, and R. B. Lindsay, Rev. Mod. Phys. **23**, 353–411 (1951)] until his monograph in 1969 [R. T. Beyer and S. V. Letcher, *Physical Ultrasonics* (Academic, New York, 1969)]. He adopted and extended new measurement techniques of ultrasonic attenuation and used them to advance our understanding of relaxation processes in liquified gases, in water and aqueous solutions, in organic liquids and organic single crystals, and in liquid metals.

9:20

3aPAb5. Bob Beyer's nontechnical contributions to acoustics. M. Strasberg (David Taylor Model Basin, NSWC, Bethesda, MD 20084-5000)

A review of Bob's contributions to the goals of our Society, including his services to the Acoustical Society of America, the American Institute of Physics, the International Commission on Acoustics, and translation journals, interspersed with anecdotes highlighting his remarkable erudition, talents, and humor.

9:35

3aPAb6. Robert T. Beyer's interaction with foreign acoustical scientists. Konstantin Naugolnykh (CIRES, Univ. of Colorado/NOAA/Envi Technol. Lab., Boulder, CO 80303)

The very essence of a scientific society is the promotion of cooperation and the exchange of information among scientists. Bob's contribution to acoustics in this field is twofold. First, he is deeply involved in the publication of acoustics literature, particularly the Russian Acoustical Journal (previously *Soviet Physics—Acoustics*, now *Acoustical Physics*). Perfect knowledge of the languages, a deep interest in the creative work, a wide range of interests, and an extensive outlook help him in this activity. Second, being openhearted and having a friendly nature, with a live interest in his colleagues, he does a lot to develop scientific links and contacts. When he visited USSR in 1958, he immediately switched on acoustical community activity, participated in the Ph.D. thesis defense of Lev Zarembo, and started a long-lived cooperation with Russian colleagues. He has educated Ph.D. and postdoctoral students from China, Norway, Russia, etc. All of them are grateful to Bob and keep a lot of good recollections. Each meeting with him leaves you with a feeling of cordial warmth and friendly concern.

9:45

3aPAb7. Robert T. Beyer's impact on the development of nonlinear acoustics. Peter J. Westervelt (Dept. of Phys., Brown Univ., Box 1843, Providence, RI 02912)

If it had not been for R. T. Beyer there would not have been papers by me and my students for JASA to reject. Why is this? He (and A. O. Williams) wrote most of my contract proposals and the associated progress reports. He and his students Stanton, Bellin, and Korman performed definitive experiments supportive of my much criticized theories. In this tribute to Beyer I will discuss these experiments and touch upon my interaction with Gene Fubini.

10:00

3aPAb8. R. T. Beyer's contributions to nonlinear acoustics. David T. Blackstock (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029 and Mech. Eng. Dept., Univ. of Texas, Austin, TX 78712-1063)

Except for a review article on radiation pressure [Am. J. Phys. (1950)], Robert T. Beyer's earliest research in nonlinear acoustics was on the increase of absorption with intensity for water and other liquids [J. Acoust. Soc. Am. (1956) (1957), and Sov. Phys.–Acoustics (1958)]. New directions soon followed: the experimental work on the parametric array [Bclli *et al.*, J. Acoust. Soc. Am. (1960) (A)], the Keck–Beyer perturbation solution for finite-amplitude waves in a viscous liquid [Phys. Fluids (1960)], and calculation of the parameter of nonlinearity B/A [J. Acoust. Soc. Am. (1960)]. Continuation and expansion of the latter work made Bob the foremost authority on B/A. Later interests include nonlinear effects in the field of a piston radiator [Ryan *et al.*, J. Acoust. Soc. Am. (1962)], finite-amplitude propagation in a relaxing fluid, self-demodulation of a pulse [Moffett *et al.*, J. Acoust. Soc. Am. (1970, 1971)], and various aspects of the crossed beams problem [J. Acoust. Soc. Am. (1970–1982)]. To these specific contributions must be added Beyer's book *Nonlinear Acoustics* and several review articles in handbooks and journals.

10:15

3aPAb9. Beyer on Beyer. Robert T. Beyer (Phys. Dept., Brown Univ., Providence, RI 02912)

Reflections.

10:30–10:45 **Break**

Contributed Papers

10:45

3aPAb10. Recollections of acoustics at Brown University in the early fifties. S. A. Elder (Phys. Dept., US Naval Acad., Annapolis, MD 21402)

When I arrived at Brown in 1950, Bob Beyer, a young assistant professor, was already becoming well known as an acoustician. Within a year he was promoted to associate professor and took over the "References" section of JASA at the death of Arthur Tabor Jones. Having previously translated works in German, soon he had taught himself Russian and become known as the editor of JETP, the Russian translation journal. I took an ultrasonics course from him, about the time the milestone article on acoustic absorption by Markham, Beyer, and Lindsay came out in Reviews of Modern Physics. In those days Brown Physics was a powerhouse of expertise in acoustics, with three out of every four faculty members specializing in that field. In 1952 distinguished acousticians from around the world gathered at Brown for an ONR ultrasonics symposium. As the field of nonlinear acoustics began to evolve, Beyer, along with Lindsay, Nyborg, Westervelt, and their students, was busy exploring high amplitude effects such as anomalous absorption, acoustic streaming, and, later, scattering of sound by sound. All this took place in an ancient relic of a building called Wilson Hall, with an attic full of dusty apparatus left from a departed 19th century department chairman, and a basement full of great, mosquito-infested ultrasonic water tanks.

11:00

3aPAb11. Vintage Beyer—Teacher, advisor, and mentor—an era of experimentation on the nonlinear interactions of sound with noise and with turbulence. Murray S. Korman (Dept. of Physics, US Naval Academy, Annapolis, MD 21402) and Timothy K. Stanton (Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

When Professor Beyer taught us ultrasonics or nonlinear acoustics at Brown, he emphasized the historical genesis of scientific ideas and included insight into the works of greats like Rayleigh and connections to Lighthill. He personalized the science. His lectures on "Radiation Pressure — The History of a Mislabeled Tensor" [J. Acoust. Soc. Am. **63**, 1025–1030 (1978)] are vintage Beyer. If our research topics were a canvas he would motivate the development of our creativity. If you asked him which color to use, you would get a philosophical story—but not which color. He provided guidance but the final brush strokes were yours. TKS and RTB's "interaction of sound with noise" (1978) featured the growth of sum and difference frequency sidebands that agreed with Westervelt. MSK and RTB's "scattering of crossed ultrasonic beams in the presence of turbulence" (1988) measured turbulent velocities. Some of Beyer's daily comments still ring through our minds "Don't just stand there — go write a thesis," or "once a graduate student — always a graduate student." However, at key talks, like our thesis defense his "When you are on the goal line shoot straight up the middle and do not attempt a long forward pass," will always echo in our minds.

11:15

3aPAb12. Beyer's *B/A* parameter and the equation of state. Bruce Hartmann, Gilbert F. Lee, and Edward Balizer (Naval Surface Warfare Ctr., Silver Spring, MD 20903)

Beyer's *B/A* parameter for liquids can be calculated using an analytical PVT equation of state supplemented with specific heat data. Results for two equations will be investigated: the Tait equation and our own. The calculations of sound speed, sound speed derivatives, and the two components of *B/A* will be compared with experimental data for a series of normal alkane liquids. A comparison of these results with the prediction of Ballou's rule (linear relation of *B/A* vs $1/c$) will be presented. Tentative conclusions about the relative utility of the two equations will be presented as well as the outlook for the approach using any PVT equation of state. [Work supported by Naval Surface Warfare Center's in-house Laboratory Independent Research Program sponsored by the Office of Naval Research.]

11:30

3aPAb13. Nonlinear acoustic Doppler effect in a bubble flow. Igor N. Didenkulov (Inst. of Appl. Phys., 46 Ulyanov Str., Nizhny Novgorod 603600, Russia), Suk-Wang Yoon, Eui-Jun Kim (Dept. of Phys., Sung Kyun Kwan Univ., Suwon, 440-746, Republic of Korea), and Alexander M. Sutin (Inst. of Appl. Phys., 46 Ulyanov Str., Nizhny Novgorod 603600, Russia)

The Doppler effect is a well-known and widely used phenomenon. In acoustics it is usually considered for linear response from a moving target. No attention was given previously to specific phenomena of nonlinear sound scattering by moving nonlinear scatterers. In the present paper a nonlinear analog of the Doppler effect is studied theoretically and experimentally. The most prominent nonlinear acoustic scatterer is a bubble. A theoretical analysis of frequency shift under nonlinear scattering of two, in a general case, acoustic waves by a moving nonlinear scatterer with generation of harmonics and combination (i.e., the sum and the difference) frequencies is given. It is shown that the nonlinear Doppler frequency shift for the difference frequency generation depends on the geometry of primary waves and can be much larger than the linear Doppler frequency shift for the difference frequency. Experiments have been made with bubble flows in a vessel. The results are in good agreement with the theory. Feasible applications of the nonlinear Doppler effect for bubble flow velocity measurement are discussed and demonstrated experimentally. [This work was done at the Acoustic Research Lab. of Sung Kyun Kwan Univ. under support of STEPI, Republic of Korea, and supported in part by RFBR (grant 94-05-16755), Russia.]

11:45

3aPAb14. Nonlinear acoustic method for bubble density measurements. Alexander M. Sutin (Inst. of Appl. Phys., 46 Ulyanov Str., Nizhny Novgorod 603600, Russia), Suk-Wang Yoon, Eui-Jun Kim (Dept. of Phys., Sung Kyun Kwan Univ., Suwon, 440-746, Republic of Korea), and Igor N. Didenkulov (Inst. of Appl. Phys., 46 Ulyanov Str., Nizhny Novgorod 603600, Russia)

An enormously high nonlinear response of a bubble to an acoustic excitation makes nonlinear methods possible and very selective for bubble observation and sizing. Among different nonlinear acoustics methods of bubble detection there is one based on difference frequency generation under the action of two high- frequency pump waves. In this method the nonlinear scattering from a single bubble was studied. In the present paper the nonlinear incoherent scattering of two high-frequency acoustic waves with the difference frequency generation from a bubble cloud was theoretically and experimentally investigated to estimate bubble size distributions. In the experiments two primary waves were produced by focused transducers. One of the primary frequencies was kept constant at 2.25 MHz; another was changed. Thus the difference frequency was varied from 30 to 320 kHz, which corresponds to the resonant bubble radii from 109 to 10 μ m. For the bubble clouds produced in a laboratory tank by electrolysis- and slit-type bubble makers, the bubble densities were well estimated with the present nonlinear acoustic method, respectively. [This work was done at the Acoustic Research Lab. of Sung Kyun Kwan Univ. under support of STEPI, Republic of Korea, and supported, in part, by RFBR (Grant 94-05-16755), Russia.]

WEDNESDAY MORNING, 15 MAY 1996 REGENCY B, 8:00 A.M. TO 12:00 NOON

Session 3aPP

Psychological and Physiological Acoustics: Binaural Hearing and Hearing Impairment

Frederic L. Wightman, Cochair
Waisman Center, University of Wisconsin, 1500 Highland Avenue, Madison, Wisconsin 53706

Marjorie R. Leek, Cochair
Audiology and Speech Center, Walter Reed Army Medical Center, Washington, DC 20307-5001

Contributed Papers

8:00

3aPP1. Effects of source spectrum irregularity and uncertainty on sound localization. Ewan A. Macpherson (Waisman Ctr., Univ. of Wisconsin—Madison, 1500 Highland Ave., Madison, WI 53705-2280)

Localization performance has been shown to deteriorate when source spectra are variable, but the few studies investigating this have confounded trial-to-trial variability with spectral irregularity present in each stimulus. In this experiment these factors were varied independently. Blindfolded listeners seated in an anechoic chamber reported the apparent locations of wideband noise bursts having either smoothly sloping spectra or 1/3-oct scrambled spectra. The spectra were either varied from trial to trial or held constant throughout an experimental session. Localization performance was compared to that obtained in a baseline condition using flat-spectrum stimuli. Despite trial-to-trial uncertainty, performance with the smoothly varying spectra was equivalent to baseline. The responses to the majority of the scrambled spectrum stimuli were similar to baseline, but pronounced bias and elevated variability were observed at certain pairings of source spectrum and location that varied between listeners. The degree of uncertainty had no substantial effect on performance with these stimuli. The results suggest that source spectrum familiarity cannot effectively compensate for the interference of spectral irregularity with the processing of spectral localization cues. However, this processing appears to be surprisingly immune to such interference over much of auditory space. [Work supported by NIDCD.]

8:15

3aPP2. Observer weighting of interaural differences of time in different regions of intracranial space. Mark A. Stellmack (Waisman Ctr., Univ. of Wisconsin—Madison, 1500 Highland Ave., Madison, WI 53705) and Raymond H. Dye, Jr. (Parmly Hearing Inst., Chicago, IL 60626)

Weights that listeners placed on interaural differences of time (IDT's) were computed for two pure tones that had different mean IDT's. As a cue on each trial, the target (753 Hz) was presented in isolation at its mean IDT. Then, the target and distractor (553 Hz) were presented simultaneously with IDT's selected from Gaussian distributions (σ = 50 μs). The listener indicated whether the target in the test interval appeared to have moved to the left or right of its cued position. The weights were the correlations between the IDT of each component and the listener's left/right response. In one condition, the mean target and distractor IDT's were of equal magnitude (0, 25, 50, 100, and 200 μs) but opposite sign. Large individual differences in weights and percent correct were observed. In a second condition, the mean target IDT was 0 μs and the mean distractor IDT was 0, 50, 100, or 200 μs. Here, target weights and percent correct generally increased for all listeners as the mean distractor IDT increased. Results are attributed to increased target weight as the difference between target and distractor IDT increased, and decreased sensitivity to changes in IDT as mean IDT increased. [Work supported by a Program Project Grant from NIDCD.]

8:30

3aPP3. Dichotic pitches as illusions of binaural unmasking. John F. Culling (Univ. Lab. of Physiol., Parks Rd., Oxford OX1 3PT, UK), Quentin Summerfield (M.R.C. Inst. of Hearing Res., University Park, Nottingham NG7 2RD, UK), and David. H. Marshall (M.R.C. Inst. of Hearing Res., University Park, Nottingham NG7 2RD, UK)

Dichotic pitches are binaural perceptual phenomena which occur when broadband noise with specific interaural phase/time relationships is presented simultaneously to the two ears. Four dichotic pitches have been described: Huggins' pitch; the binaural edge pitch; the Fourcin pitch; the dichotic repetition pitch (DRP). This paper shows that the perceived pitch and timbre of the first three dichotic pitches are correctly predicted using a model of binaural unmasking [Culling and Summerfield, J. Acoust. Soc. Am. **98**, 785–797 (1995)]. The model detects across-frequency variations in interaural correlation; the resulting central spectrum shows peaks which correspond to the frequencies listeners hear. These pitches are probably, therefore, illusions produced by the mechanism of binaural unmasking. The DRP is not predicted by the model. The DRP could be produced through binaural cross talk, but experiments using insert earphones which produce minimal cross-talk confirmed that the phenomenon persists when cross talk is effectively eliminated. The DRP is probably, therefore, produced by a separate mechanism from the other dichotic pitches.

8:45

3aPP4. Binaural detection with spectrally nonoverlapping signals and maskers. Marcel van der Heijden, Constantine Trahiotis [Surgical Research Ctr., Dept. of Surgery (Otolaryngol.) and Ctr. for Neurol. Sciences, Univ. of Connecticut Health Ctr., Farmington, CT 06030], Armin Kohlrausch, and Steven van de Par (Inst. for Perception Res., NL-5600 MB Eindhoven, The Netherlands)

Thresholds were measured for diotic tonal signals masked by interaurally delayed bands of Gaussian noise. When the signal frequency was 525 Hz, the spectrum of the masker was either below (highest frequency, 450 Hz) or above (lowest frequency, 600 Hz) the frequency of the signal. With a 450-Hz signal, the spectrum of the masker was always above the signal

3a WED. AM

frequency (lowest frequency, 600 Hz). Signals had a 250-ms duration and were temporally centered within 300-ms-long maskers. The spectrum level of the maskers was 60 dB. Masked thresholds obtained in all three conditions varied sinusoidally with the interaural delay of the masker. The periodicities within the data were close to, but not identical with, the periodicities of the *signals* (not the maskers). This outcome is discussed in terms of (1) auditory distortion products stemming from interactions within the masking bands of noise [cf. van der Heijden and Kohlrausch, J. Acoust. Soc. Am. **98**, 3125–3134 (1995)], and (2) binaural models incorporating internal delay mechanisms. It is shown that using such spectrally nonoverlapping signals and maskers can help test interesting hypotheses concerning binaural models. [Work supported by Grant No. 5 RO1 DC00234-13 from the National Institute of Deafness and other Communication Disorders, NIH.]

9:00

3aPP5. Lateral position of dichotic pitches can be substantially affected by interaural intensitive differences. Anthony N. Grange and Constantine Trahiotis (Surgical Res. Ctr., Dept. of Surgery (Otolaryngol.) and Ctr. for Neurol. Sciences, Univ. Ctr. of Connecticut Health Ctr., Farmington, CT 06030)

Three listeners used an acoustic pointer to match the intracranial position of dichotic pitches presented with interaural intensitive differences (IIDs). Unlike data reported previously [Raatgever, "Binaural time processing and time-intensity trading," in *Psychophysics, Physiology, and Behavioural Studies in Hearing*, edited by G. van den Brink and F. A. Bilsen (Delft U.P., Delft, The Netherlands, 1980), pp. 425–453], the intracranial position of the dichotic image was moved substantially by IIDs. Large effects of IIDs were found in a variety of stimulus conditions, including one investigated by Raatgever. When the dichotic stimuli were not pulsed, but were presented continuously and turned off only when the acoustic pointer was presented, one of the listeners produced data like those previously reported by Raatgever. It appears that intracranial images produced by stimuli which support dichotic pitches, like images produced by other binaural stimuli, *can* be affected substantially by IIDs. Consequently, in terms of lateralization, it does not appear necessary to maintain a theoretical cleavage between images produced by dichotic pitches (previously thought to be influenced only by ITDs) and images produced by other binaural stimuli. [Work supported by Grant No. 5 RO1 DC00234-13 from the National Institute on Deafness and other Communication Disorders, NIH.]

9:15

3aPP6. Localization and speech perception in noise by aging listeners. Robert C. Bilger (Dept. Speech and Hearing Sci., Univ. of Illinois, 901 S. Sixth St., Champaign, IL 61820), Chien Yeh Hsu (Inst. for Information Industry, Taipei, Taiwan, Republic of China) and Ted A. Meyer (Washington Univ., St. Louis, MO 63110)

Presbycusic listeners are reported to have difficulty hearing speech in noisy sound fields. To resolve the issue of whether this difficulty is central or peripheral, an experiment, in which the spectrum of a masking noise was manipulated to replace acoustics that a listener's peripheral hearing loss filtered out of the input, was conducted. The experimental listeners were 12 people with high-frequency hearing loss who were 65 years of age or older (5 women/7 men) and the controls were 12 young adults with normal hearing (8 women/4 men). When PB words (200 words/condition) were presented at 65 dB SPL in a flat noise ($S/N = 0$ dB) at zero azimuth, the controls scored 53.9% correct and, when the speech and noise were separated by 90°, their score was 70.2% correct, an increase of 16.3% Under the same experimental conditions, the experimental listeners scored 44.8% correct for signal and noise at zero azimuth and 53.4% correct for 90° separation, an increase of only 8.6% When the noise presented to the experimental listeners was filtered to match their equal-loudness contours (high-frequency boost), however, their speech recognition improved to 59.5%, an increase of 14.7% over their mean score for no separation of signal from noise, evidence that their difficulty is of peripheral origin. [Work supported by DC 00174-11.]

9:30

3aPP7. Detection and identification of environmental sounds. Charles S. Watson, Gary R. Kidd, and Brian Gygi (Dept. of Speech and Hear. Sciences, Indiana Univ., Bloomington, IN 47405)

Most studies of auditory recognition and identification have employed either speech stimuli or nonspeech sounds generated in the laboratory (e.g., tones of various frequencies, tonal patterns, click trains). The present study employed 25 naturally occurring complex sounds (obtained from a commercial sound-effects library), such as those produced by doors closing, babies crying, helicopters in flight, and other familiar events. These sounds, equated for peak levels, were recorded with a background of broadband noise. The recorded sounds were presented to groups of six to eight listeners in both open- and closed-set formats (with the list of responses displayed continuously in the latter). Confusion matrices were generated using a wide range of event-to-noise (Ev/N) ratios. Two frequent confusions were identified for each item, and were used to create a three-alternative forced-choice test. Eight values of Ev/N were selected for each item in an effort to achieve uniform item identifiability. Average Ev/N required to achieve 50-percent-correct recognition, the slopes of psychometric functions, and the distribution of these measures among normal-hearing listeners are compared to the corresponding measures for speech identification. [Work supported by AFOSR and NIDCD.]

9:45

3aPP8. Perceptual fusion of multiple echoes. Sandra J. Guzman and William A. Yost (Parmly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

In a natural environment, perceptual fusion of a sound may occur despite the fact that multiple echoes exist. If a train of repeating sources and echoes is established experimentally, the echoes are suppressed and perceptual fusion is established. In the present study, perceptual fusion was investigated by having listeners attend to the last sound presented in a train of sources and echoes. The effects of the number of sounds presented during a train, time delay between sources and echoes, relative amplitude between sources and echoes, and number of echoes presented were investigated. Results indicate that there is an interaction between time delay and number of echoes in perceptual fusion. Relative amplitude of the source and echoes also affected perceptual fusion. These findings will be discussed in terms of the conditions under which perceptual fusion seems to occur and its relation to a listener's prior experience in an acoustic environment. [Work supported by NIDCD and AFOSR.]

10:00–10:15 Break

10:15

3aPP9. Franssen effect as a function of carrier waveform, carrier level, and room acoustics. Dan Mapes-Riordan, William A. Yost, and Sandra J. Guzman (Parmly Hearing Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

The Franssen effect was studied as a function of the frequency of the source and echo sounds, the difference in frequency and level between the source and echo sounds, and as a function of the amount of room reverberation. A modification of the single-interval procedure used by Hartmann and Rakerd [J. Acoust. Soc. Am. **86**, 1366–1373 (1989)] was used to measure the degree to which the source and echo sounds were fused in terms of determining the location of the perceived sound. The amount of fusion was a nonmonotonic function of frequency, decreased as the frequency and level difference between the echo and source increased, and is greater for a room with more reverberation. The results will be discussed in terms of the precedence effect, listener's prior experience, and the plausibility hypothesis. [Work supported by a Program Project Grant from NIDCD and a grant from the AFOSR.]

10:30

3aPP10. Target weights for high-frequency SAM targets and distractors lateralized on the basis of envelope delay. R. H. Dye (Parmly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

The extent to which judgments of the laterality of SAM high-frequency carriers is influenced by other SAM tones was assessed in a stimulus-classification paradigm. The target was either a 4- or a 2-kHz tone modulated at 200 Hz, and the distractor was a 2- or 4-kHz tone modulated at rates ranging from 50 to 400 Hz. Of particular interest is the manner in which the weight given to the target is dependent upon the carrier and modulation frequencies. On each trial, the target was presented with one of ten envelope delays, as was the distractor. Each test interval was preceded by a diotic presentation of the target alone. The relative contributions of the envelope delays of the target and the distractor were assessed by the point-biserial correlations between the target/distractor delays and the type of response elicited ("left"/"right"). Although substantial individual differences were found in terms of which carrier was more potent and in how target weight depended upon modulation frequency, the relationship between target weight and modulation frequency tended to be the same for the two carrier frequencies for a given listener.

10:45

3aPP11. Phase discrimination and stimulus level in normal-hearing and hearing-impaired listeners. Marjorie R. Leek (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001), Roy D. Patterson (MRC Appl. Psych. Unit, Cambridge CB2 2EF, UK), and Van Summers (Walter Reed Army Medical Ctr., Washington, DC 20307-5001)

In a linear filterbank there is a simple trade-off between filter bandwidth and duration of the impulse response. This suggests that hearing-impaired listeners with broad auditory filters will have abnormally acute temporal sensitivity, and would be more sensitive than normal to phase manipulations in harmonic stimuli. However, repeated attempts to substantiate this prediction have largely failed. Linear models fail to account for the loss of gain associated with outer haircell damage in the impaired cochlea. In preparation for modeling phase discrimination with an active cochlea model, phase discrimination thresholds were measured at a number of stimulus levels for normal-hearing and hearing-impaired listeners in a standard/2AFC experiment. Stimuli were 21-component harmonic complexes with a fundamental frequency of 125 Hz. The standard had all components in cosine phase. Test stimuli had odd-numbered harmonics in cosine phase and even-numbered harmonics shifted a fixed number of degrees as determined by an adaptive track. For both groups of listeners, phase discrimination deteriorated as stimulus level decreased below 30–40 dB SL. The importance of sensation level to these results and their lack of dependence on frequency resolution will be discussed in terms of linear and level-dependent models of cochlear function. [Work supported by NIH.]

11:00

3aPP12. A comparison of the effects of active noise reduction and conventional hearing protectors on auditory perception in normal and hearing-impaired listeners. Sharon M. Abel and Deborah L. Spencer (Dept. of Otolaryngol., Mount Sinai Hospital, 600 University Ave., Toronto, ON M5G 1X5, Canada)

The relative benefit of active noise reduction (ANR), compared with conventional hearing protection, was assessed. Two groups of normal-hearing subjects, differing in age, and one group with bilateral sensorineural hearing loss participated. Subjects were tested with the ears unoccluded, and fitted with E-A-R foam plugs, E-A-R HI-FI plugs, Bilsom Viking muffs, and Peltor 7004 muffs with ANR capability. The E-A-R foam plug provided the highest, and the E-A-R HI-FI plug the lowest, real-world sound attenuation, from 250 Hz to 8 kHz. The Bilsom and Peltor muffs were midway and virtually identical. ANR reduced sound levels by an additional 10 dB at 250 Hz. In normal listeners, protectors improved word recognition in noise but there was no difference due to the device. In impaired listeners, performance in quiet was significantly better with the ears unoccluded or fitted with the E-A-R HI-FI plugs than with the other conventional devices. ANR was midway. In noise, performance was relatively best with ANR and worst with the ears unoccluded. Within group, there were no differences in either duration of frequency difference limens, across conditions. Protected frequency DLs were significantly greater for the impaired group. The results support considering both hearing status and task in assessing hearing protectors. [Work supported by National Defence Canada.]

11:15

3aPP13. Differences in modeling adaptation for the hard of hearing versus normal listeners. Hong-Wei Dou, Ian Mackay, Maureen Korman, Teri Huber, Marianne Martin, Heather Bose, Stephanie Hall, Dana Homan, Gena Walls, Dana Hawes, David Sandman, and Ernest M. Weiler (Communication Sciences and Disord., Univ. of Cincinnati, ML No. 379, Cincinnati, OH 45221)

Data gathered by Korman and Weiler [MA thesis, Univ. of Cincinnati, 1986] and Janson *et al.* [Br. J. Audiol. **30**, 35–42 (1996)] have shown the loudness function for loudness adaptation gathered by the ipsilateral comparison procedure (ICP) differs for normal hearing listeners and those with high-frequency cochlear loss. This paper investigates the modeling assumptions for loudness growth in the two kinds of listeners. Surprisingly, the loudness growth within a given frequency is not clearly different in kind, although the two groups start with different values in most test conditions. However, there does appear to be a difference across frequencies which may be a reflection of the spread of excitation on the basilar membrane. Expected differences in activation for normal listeners and those with a high-frequency loss will be discussed.

11:30

3aPP14. Multimicrophone directional hearing aid using hybrid adaptive beamforming. G. L. Gibian W. S. Koroljow (Planning Systems, Inc., 7923 Jones Branch Dr., McLean, VA 22102)

Recent studies have demonstrated some success in improving speech intelligibility when either competing voices or reverberation are present, using an adaptive beamformer in the first case and a fixed-weight beamformer in the second. The present study evaluates the effectiveness of a hybrid beamformer incorporating the strengths of an adaptive beamformer with the advantages of an optimally chosen fixed-weight beamformer. Received signals at a short array of microphones in a room were simulated for a single talker and for a babble of voices at each of several interferer locations. Three rooms with different absorption coefficients were simulated. Conditions with more than one simultaneous interferer location and with starts and stops of an interferer are being tested. Success of the hybrid adaptive beamformer is being assessed by informal listening and by measuring the intelligibility-weighted improvement in SNR. Preliminary results appear promising.

11:45

3aPP15. Predicting custom hearing aid circuit selection success from field returns analysis. David A. Preves and Mary E. Leisses (Argosy Electronics, 10300 W. 70th St., Eden Prairie, MN 55344)

A wide variety of signal processing circuitry is available from hearing aid manufacturers for use in custom hearing aids. Whether the type of circuit used is specified by the clinician or by the manufacturer, the selection process is often speculative because of the lack of understanding about how different types of signal processing interact with pathological auditory systems. An empirical approach to signal processing selection is proposed based on the circuit that is most likely to minimize field returns for a particular audiometric configuration, degree of loss, and hearing aid model. Several thousand ITE and ITC hearing aid fittings made by one hearing aid manufacturer were monitored via a computerized field tracking data base for incidence of repairs and returns for credit. Circuits included four types of signal processing having class D output stage: linear, input compression, TILL, and BILL. Audiograms were divided into seven configurations: flat, gradually sloping, steeply sloping, precipitous, rising, cookie bite, and inverted cookie bite. The database was entered into the IDIS artificial intelligence program, and design rules were formulated to maximize fitting success. From these rules, signal processing type(s) were identified for each audiometric configuration that would minimize field returns.

WEDNESDAY MORNING, 15 MAY 1996 CANYON HALL, 8:00 TO 10:30 A.M.

Session 3aSAa

Structural Acoustics and Vibration: Plates, Beams, and Shells

Sabih I. Hayek, Cochair
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John A. Burkhardt, Cochair
Department of Engineering, Indiana-Purdue University at Ft. Wayne, 2101 East Coliseum Boulevard, Ft. Wayne, Indiana 46805-1499

Contributed Papers

8:00

3aSAa1. Application of matched-field processing to structural vibration problems. Gabriella Turek (Marine Physical Lab., Scripps Inst. of Oceanogr., 8603 B La Jolla Shores Dr., La Jolla, CA 92093-0238) and W. A. Kuperman (Scripps Inst. of Oceanogr., San Diego, CA 92093-0701)

The applicability of matched-field processing (MFP) [Baggeroer *et al.*, IEEE J. Ocean. Eng. **18**, 401–424 (1993)] techniques to localize sources of vibration in structures and to perform nondestructive testing is explored. MFP is a generalized procedure of array processing used in ocean acoustics to either localize sources or perform inversions. MFP involves correlations between the solutions (or "replicas") of the wave equation for a given acoustic model of the ocean and the data measured at an array of sensors. The correlations are made using an assortment of linear and nonlinear methods. These techniques were successfully tested using simulation for the relatively simple problem of localizing a harmonic point force on a simply supported laterally vibrating beam. Wave equation solutions for describing the vibration of more complicated structures (simulated by adding spring constraints to the beam) become, at some point of complexity, intractable. Often statistical techniques provide the only practical description of such structural vibration problems. Multiple constraint MFP methods, tolerant to solution mismatch, were used to deal with these more complicated structures. Finally, it was demonstrated that MFP methods might also be used as a means of nondestructive testing in order to locate defects.

8:15

3aSAa2. The phase gradient method applied to the plate: Analysis of the partial derivatives with respect to the phase velocities. O. Lenoir, J. M. Conoir, and J. L. Izbicki (Lab. d'Acoust. Ultrason. et d'Electron. LAUE, URA CNRS 1373, Univ. du Havre, Pl. Robert Schuman, 76610 Le Havre, France)

The phase gradient method is a convenient tool in order to analyze the frequential and angular resonances of an immersed elastic plate [J. Acoust. Soc. Am. **94**, 330–343 (1993)]. It consists in the study of the derivative of the phase of the reflection coefficient with respect to the frequency x—the obtaining of the resonance frequency x^* and the resonance width Γ^* is then straightforward—or with respect to the angular parameter y (which is the sine of the incidence angle). It is shown that the derivatives with respect to the different phase velocities involved in the scattering problem also permit the isolation of the resonances. The derivatives with respect to the longitudinal (c_L) and shear (c_T) velocities of the plate and to the longitudinal velocity of the fluid (c_F) are studied. The plot of the previous derivatives, versus the frequency, exhibits Breit–Wigner peaks. Their half-widths are equal to Γ^* and the coefficients Γ_L, Γ_T, Γ_F related to their amplitude are introduced. The physical meaning of the previous amplitude coefficient is given. Then a resonant energy conservation law is shown: $\Gamma^*+\Gamma_F=\Gamma_L+\Gamma_T$.

8:30

3aSAa3. Waves and vibrations in multilayer poroelastic plates. M. Cengiz Dökmeci (Istanbul Tech. Univ.–Teknik Üniv. P.K. 9, Taksim, 80191 Istanbul, Turkey) and G. Askar Altay (Boğaziçi Univ., Bebek, 80815 Istanbul, Turkey)

In view of a recent extensive review concerning wave propagation in porous media [M. Y. Corapcioğlu, in *Transport Processes in Porous Media* (Kluwer, Dordrecht, 1991), pp. 373–469], this paper presents a theory for the dynamic response of a multilayer plate within the frame of Biot's theory of elasticity and consolidation for a porous anisotropic solid [M. A. Biot, J. Appl. Phys. **26**, 182–185 (1955)]. The multilayer plate may comprise any number of bonded layers, each with a distinct but uniform thickness and anisotropic elastic properties. A differential variational principle [G. A. Altay and M. C. Dökmeci, J. Acoust. Soc. Am. **95**, 3007 (A) (1994)] for Biot's theory together with Mindlin's plate kinematics for each layer is used to derive the two-dimensional approximate equations of multilayer plate in invariant differential and variational forms. All the continuity conditions between the interfaces of layers are taken into account. The governing equations are capable of predicting the extensional, flexural, torsional, and coupled vibrations of multilayer poroelastic plates. Special cases involving the geometry, material, and motions of multilayer plates are studied. The results contain some of earlier ones as special cases. [Work supported in part by TÜBA-TÜBİTAK.]

8:45

3aSAa4. Mode shape and eigenfrequency spectra of elastic plates in the vicinity of the thickness resonance frequency. Victor T. Grinchenko (Inst. of Hydromech., Natl. Acad. of Sciences, Kiev 252057, Ukraine)

A number of experimental investigations have been performed to study vibration characteristics of elastic plates in a high-frequency region. Up to this time the empirical data have no satisfactory explanation in terms of Lamb wave coupling. The theoretical analysis of the coupling effects presented in the article is based on the exact solution of the free-axis-symmetric vibration problem for a circular plate. Consideration of a special case of zero Poisson's ratio allows one to distinguish and investigate three independent kinds of eigenmodes. Two of them are initiated by simply propagated longitudinal waves. The waves with complex wave numbers (evanescent standing waves) provide a rise to a third type of eigenform. Eigenfrequencies of these last modes are very close to the thickness resonance frequency and they increase as radius of a plate increases. The data for nonzero Poisson's ratios demonstrate an influence of coupling on a structure of eigenfrequency spectra and mode shape. An investigation of radius variation effects gives a good indication of the feasibility of the coupling effect control. The values of the plate radius with the moderate mode shape disturbances are determined.

9:00

3aSAa5. Investigation of wave processes in a Timoshenko thin-walled beam of open section caused by the shock subjection of an elastic sphere upon its surface. Yuriy A. Rossikhin (Dept. of Theoret. Mech., Voronezh State Acad. of Construction and Architecture, ul. Kirova 3-75, Voronezh 394018, Russia)

The problem on the normal impact of an elastic sphere upon an elastic Timoshenko beam of thin-walled monosymmetrical open section is considered. The process of impact is accompanied by the dynamic flexure and torsion of the beam, resulting in the propagation of plane flexural, shear, and torsional waves of strong discontinuity along the beam axis. Behind the wavefronts the solution is constructed in terms of one-term ray expansions. During the impact the sphere moves under the action of the contact force which is determined due to the Hertz's theory, but the contact region moves under the action of the contact force, and the bending-torsional moment and transverse forces, which are subjected to the lateral surfaces of the contact region and are determined using one-term ray expansions. The joint consideration of the equations of sphere and the contact region motion leads to the Abel equation of second kind in the rate of change of the value of the sphere and beam drawing together. The solution to the Abel equation written in the form of a series in terms of this value allows one to determine all characteristics of the sphere and beam shock interaction.

9:15

3aSAa6. Wave-number-based assessment of the doubly asymptotic approximation. Jerry H. Ginsberg and Kuangchung Wu (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The most widely implemented technique for modeling fluid–structure interaction effects associated with shock response is the doubly asymptotic approximation (DAA), which has been developed in a variety of versions. Because of a lack of analytical solutions for realistic geometries, prior validation efforts have not provided definitive guidelines regarding the accuracy of the method. The present work uses Nicholas–Vuillierme's derivation [*Numerical Techniques in Acoustic Radiation*, edited by R. J. Bernhard and R. F. Keltie (American Society of Mechanical Engineers, New York, 1989), Vol. 6, pp. 7–13] of the frequency-domain version of DAA as the basis for examining the accuracy and limitations of DAA for a slender hemi-capped cylindrical shell. The development computes the wet surface impedance matrix relating surface pressure and velocity variables over a broad frequency band according to two DAA versions and the surface variational principle. The alternative wet surface impedances are used to predict the frequency-domain structural response of the shell to a point force, after which alternative time-domain responses are obtained from the FFT algorithm. Results for wave-number amplitudes (azimuthal and meridional) are reported for moderate and large aspect ratios. An overview indicates that a higher-order DAA version occasionally works extremely well, but that there is no consistent high/low trend as a function of wave number. [Work supported by the Office of Naval Research, Code 1222.]

9:30

3aSAa7. Robust feedback control of flow-induced structural radiation of sound. Craig Heatwole, Matthew A. Franchek, Robert J. Bernhard, and Luc Mongeau (1077 Ray W. Herrick Lab., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

A significant component of the interior noise of aircraft and automobiles is a result of the turbulent boundary layer excitation of the vehicular structure. In this investigation the feasibility of active control of wind noise is studied. Initial studies are made by considering the idealized case of sound radiation from a simply supported panel excited by a turbulent flow boundary layer. To study various control system alternatives, a wind noise model is developed. The model consists of a modal model of the excitation of a plate from a turbulent boundary layer, a model of plate vibrations, and a model of the sound radiated from the plate. A frequency-domain approach, similar to quantitative feedback theory, is utilized in the design of a robust feedback controller. The controller utilizes plate acceleration feedback while maintaining sound-pressure-level reduction as the primary design objective. The design method insures that performance and control effort objectives are met for plant and disturbance uncertainty. The controller design method offers flexibility in configuring the size of the controller, and the effect of the performance and control effort objectives on the controller design are easily identified.

9:45

3aSAa8. Adaptive control of modular sound enclosure. Kyle Martini (Cambridge Acoustical Assoc., 200 Boston Ave., Medford, MA 02155) and Ann Stokes (Boston Univ., Boston, MA)

This paper presents an experimental demonstration of the active control of a novel, modular sound enclosure that is built of panels which are stiffness controlled throughout the frequency range of interest. In this range, traditional passive noise enclosures are difficult to successfully implement. The design of the lightweight, stiff panels permits active control with a manageable number of actuators and sensors. The control technique is the adaptive, feedforward, filtered-x approach. Adaptive control is needed because realistic changes in the enclosure temperature and structural resonances would degrade the performance of fixed feedforward filters. The design of the enclosures makes this adaptive approach computationally feasible. Two performance criteria are evaluated: the minimization of the sum of the squares of the accelerations of the panels, and the minimization of certain supersonic wave-number components of the surface vibration field. Both approaches are straightforward for this enclosure design, where each panel behaves like a piston. [Work supported by the National Science Foundation.]

10:00

3aSAa9. An experimental study of active control of vibration transmission in a cylindrical shell. Xia Pan and Colin H. Hansen (Dept. of Mech. Eng., Univ. of Adelaide, Adelaide, South Australia 5005, Australia)

An experimental investigation of active control of vibratory power transmission in a semi-infinite cylinder, using an array of electrodynamic shakers, is described. The cylinder was mounted vertically in a sand-filled box (1 m in height) in an attempt to provide a semianechoic termination at one end. The other end was simply supported using a steel ring. The cylinder was excited near the simply supported end using a primary shaker. Control was implemented using three control shakers applied downstream from the primary shaker. Using three control shakers it is possible to achieve a 7-dB reduction in power transmission. Although 7 dB was the maximum power reduction which could be obtained for the experimental equipment, the numerical simulation shows that on a large scale, a reduction of 30 dB could be obtained. Also, measurement of power transmission intensity in a cylinder using two accelerometers was investigated. The results indicate a means of simplifying power transmission measurements in large cylindrical structures.

10:15

3aSAa10. A general dynamic theory of thermopiezoceramic shells. G. Aşkar Altay (Dept. Civil Eng., Boğazici Univ., Bebek, 80815 Istanbul, Turkey) and M. Cengiz Dökmeci (Istanbul Tech. Univ.–Teknik Univ., Taksim, 80191 Istanbul, Turkey)

This study presents a general theory for the motions of a ceramic shell in which there is coupling among mechanical, electrical, and thermal fields. The coated ceramic shell is treated as a two-dimensional thermopiezoelectric medium and a separation of variables solution in terms of the thickness coordinates, the midsurface coordinates, and time is sought for its field variables. Then, a variational averaging procedure [M. C. Dökmeci, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **35**, 775–787 (1988)] together with the solution is used so as to derive the system of approximate equations of ceramic shell. The invariant system of governing equations which are expressed in both differential and variational forms accounts for all the types of motions of ceramic shells. Certain cases involving special geometry, material properties, and motions are considered [e.g., M. C. Dökmeci, J. Math. Phys. **19**, 109–126 (1978)]. Also, the sufficient boundary and initial conditions are given for the uniqueness in solutions of the fully linearized system of shell equations. The results are shown to generate a series of known shell theories [e.g., M. C. Dökmeci, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **37**, 369–385 (1990) and references therein]. [Work supported by TÜBA-TÜBİTAK.]

WEDNESDAY MORNING, 15 MAY 1996 CANYON HALL, 10:45 A.M. TO 12:00 NOON

Session 3aSAb

Structural Acoustics and Vibration: Radiation and Damping

Gerard P. Carroll, Chair
Carderock Division, Naval Surface Warfare Center, Bethesda, Maryland 20084-5000

Contributed Papers

10:45

3aSAb1. Specific damping capacity for high-loss materials. Gilbert F. Lee and Bruce Hartmann (Naval Surface Warfare Ctr., Silver Spring, MD 20903)

The purpose of this presentation is to establish the relation between specific damping capacity and loss factor for high-loss materials. The motivation for this work is that the commonly used relationship between these variables is only valid for low-loss materials and leads to significant errors as the loss increases. The relationship cited in numerous references between specific damping capacity is $\Delta W/W = 2\pi \tan\delta$. This equation is only valid for very small values of δ. Another difficulty with the above relationship is that the specific damping capacity is generally assumed to represent the fraction of energy converted to heat. For large values of tan δ, $\Delta W/W$ can be greater than one, which seems to be a contradiction. In this presentation, a relationship between specific damping capacity and loss factor is derived, which is valid for all values of tan δ. In the limit as the loss factor approaches infinity, the specific damping capacity approaches unity. A comparison of our relationship with others found in the literature will be presented. [Work supported by Naval Surface Warfare Center's In-house Laboratory Independent Research Program sponsored by the Office of Naval Research.]

11:00

3aSAb2. Precise formulation of the vibroacoustic effects of an added constrained damping layer on a plane structure. Jean Nicolas, Noureddine Atalla, Olivier Foin, and Bertrand Mercier (Mech. Eng. Dept., Univ. de Sherbrooke, Sherbrooke, PQ J1K 2R1, Canada)

The model proposed here describes the SAV behavior of a sandwich plate (elastic–visoelastic–elastic) and uses the following hypothetisis, inspired by the discrete layer theory: The viscoelastic layer takes into account bending, shear, and traction compression; normal strain and torsional motion are neglected; using the continuity of the displacement at the visoelastic layer interfaces, the displacement field of the viscoelastic layer is written as a function of the elastic layer displacement field. A variational method is used to find the displacement equations. The radiation impedance is calculated via a formulation recently proposed by Atalla and Nicolas [J. Acoust. Soc. Am. (1994)]. This formulation allows one to calculate, for a large frequency range, the quadratic velocity, the radiation efficiency, and the radiated power. Emphasis is then put on precise experimental validation. A fully instrumented setup using a laser vibrometer for probing the displacement field has been developed. The basic plate is made either from aluminum or from resin epoxy. Using technical data furnished by the manufacturers, the comparison between theory and experience is extremely good in the case of aluminum. Discrepancies are bigger for the resin epoxy and this aspect will be discussed. Partial covering, which is also of great practical interest, will also be presented for both theoretical and experimental aspects.

11:15

3aSAb3. Internal–external measurement of three-dimensional velocities of a submerged shell by optical vibrometry. Joseph Vignola, P. Frank, L. Leka (SFA, Landover, MD 20785), Harry Simpson, and Brian H. Houston (Naval Research Laboratory, Washington, DC 20375-5000)

A unique experimental capability for measuring both the interior and exterior three-axis surface motions of a completely submerged ribbed cylindrical shell has been developed. This system is composed of two three-dimensional laser Doppler vibrometers positioned by a pair of high-precision robots. The two vibrometers work together to measure the three-components of velocity up to 50 kHz on both the interior and exterior cylinder surfaces. For the interior, a compact three-dimensional vibrometer was designed and constructed to operate in conjunction with a cylindrical-based scanner mounted to the stiff endcaps of the shell. This scanning system is capable of positioning the laser vibrometer with a precision of +/− 3 mil over 100% of the interior cylinder surface. A second three-dimensional laser vibrometer designed for exterior (in-water) operation was constructed and integrated into the NRL near-field acoustic holography system. This large workspace Cartesian-based scanner is capable of collocating the focal point of the exterior vibrometer with that of the interior to within +/− 10 mil. The design of the system will be discussed, and structural acoustic data will be presented from a series of preliminary experiments conducted to evaluate the performance of the system.

11:30

3aSAb4. Acousto-optic phase modulation of a laser Doppler vibrometer signal resulting from the radiated pressure produced by a submerged cylindrical shell: Numerical predictions and experimental results. Harry J. Simpson, Joseph Vignola, Martin H. Marcus, and Brian H. Houston (Naval Res. Lab., Code 7136, 4555 Overlook Ave., Washington, DC 22375)

It has been proposed that large stand-off laser Doppler vibrometer (LDV) measurements have significant errors resulting from index of refraction modulation due to the acoustic pressure distribution along the laser light paths between the structure and the probe [Y. H. Berthelot *et al.*, J. Acoust. Soc. Am. **97**, 3347(A) (1995)]. A high fidelity numerical model has been used to predict the pressure distribution and shell motions for a submerged finite ribbed cylinder under point excitation. Generally speaking, these predictions show low levels of contamination even for large stand-off probe distances. In addition, an experimental study has been conducted at NRL using a newly integrated three-axis vibrometer and near-field pressure probe where shell motions and near-field pressure distributions were measured for the case of a submerged ribbed cylinder under point excitation. The experiments and the associated numerical modeling both indicate low levels of acousto-optic contamination of the shell velocity signals. These findings contrast previously reported predictions of the level of acoustically induced error for large stand-off in-water LDV designs. [Work supported by ONR.]

11:45

3aSAb5. Fractal sound radiation by layered elastic structures. Leonid M. Lyamshev (N. N. Andreev Acoust. Inst., Shvernik Ste. 4, 117036 Moscow, Russia)

A general method of calculation of sound radiation by layered elastic structures is developed. The reciprocity theorem formulated by the author in 1958 is used. It is shown that plane structure sound radiations are determined by external acting forces and the structure transmission coefficient. The method is very convenient for the analysis and calculation since the external force characteristics are given and the transmission coefficient is a known function of impedances of layers. Several particular problems are considered. Sound radiation by a thin plate or a thick three-layered structure are considered when concentrated or fractal distributed statistical external forces act. The method for moving media in layers is generalized. Sound radiation by curved inhomogeneous layered structures is discussed. The plane structure formula, Green's theorem, and the Kirchhoff approximation are used in last cases. Sound radiation fractal characteristics are discussed.

WEDNESDAY MORNING, 15 MAY 1996 — REGENCY D, 8:15 A.M. TO 12:15 P.M.

Session 3aSC

Speech Communication and Musical Acoustics: Speech and Music: Exchange of Ideas, Methods and Findings

Barbara Acker, Cochair
Center for Cognitive and Psycholinguistic Sciences, Binghamton University, Binghamton, New York 13902-6000

Richard E. Pastore, Cochair
Psychology Department, State University of New York, Binghamton, New York 13902-6000

Chair's Introduction—8:15

Invited Papers

8:25

3aSC1. Perceiving and encoding timbre. Mark A. Pitt (Dept. of Psychol., 1885 Neil Ave. Mall, Ohio State Univ., Columbus, OH 43220)

Interactions with the auditory environment often require accurate recognition of sound objects and events (e.g., voices and instruments, and the utterances that emanate from them). Two lines of research in the lab have been exploring how listeners process the auditory dimension of timbre as one step toward understanding object recognition. In one, commonalties in processing different classes of timbres (human voices, musical instruments) using the selective adaptation paradigm are explored. In the second, the characteristics of timbre memory using the interpolated tone paradigm are examined [Deutsch, Science **168**, 1604–1605]. Findings will be discussed in the context of similar results in the literature.

8:50

Commentary by Paul Iverson and Michael D. Hall
Dept. Speech and Hear. Sciences, Univ. of Washington, Seattle, WA 98195

9:05–9:15 Discussion

9:15

3aSC2. Perceptual anchors and magnets in infancy. Sandra E. Trehub (Univ. of Toronto, Erindale Campus, Mississauga, ON L5L 1C6, Canada)

Research on adults' and children's perception of tone sequences or melodies reveals that they more readily detect the same magnitude of change in the context of a "good" or well-formed sequence than in the context of a "poor" or less well-formed sequence. Because well-formedness is typically confounded with familiarity, the origin of such performance asymmetries often remains unclear. In recent studies, however, 6-month-olds exhibited adult-like patterns of performance, raising the possibility of processing predispositions for good auditory sequences. Such sequences could be considered "natural" prototypes, and would be expected to occur frequently across cultures. Natural prototypes might serve a perceptual anchoring function for novice and experienced listeners, facilitating the acquisition of certain kinds of information. Kuhl argues, however, that phonetic prototypes operate differently, being less discriminable from other sounds within the same category than are nonprototypes. In other words, the good instances function as magnets rather than anchors. On the one hand, it is useful for natural prototypes to function as anchors, being highly distinctive and memorable. On the other hand, sufficient flexibility is necessary for the acquisition of culture-specific or "magnetic" prototypes.

9:40–9:55

Commentary by Peter W. Jusczyk
Dept. of Psych., SUNY at Buffalo, Buffalo, NY 14260

9:55–10:05 Discussion

10:05–10:20 Break

10:20

3aSC3. Musical communication and theories of the stimulus. Caroline Palmer (Dept. of Psychol., 1885 Neil Ave. Mall, Ohio State Univ., Columbus, OH 43220)

Is the stimulus for music perception an abstract form of a musical piece, or a particular performance of it? As with language, music reflects a communication of structure among its users (composers, performers, and listeners), which suggests the existence of some shared musical knowledge. Music performance may be an indispensable source of evidence for theories of that shared knowledge, because the particular expression given in each performance guides our musical understanding. Different sources of knowledge in music performance that contribute to a theory of the stimulus for perception will be described. Evidence from both expressive nuances of performance and from production errors (unintended mistakes) converge on the same knowledge sources. These performance findings have correlates in both music perception and in other production domains, such as speech. [Work supported by NIMH and NSF.]

10:45–11:00

Commentary by William Forde Thompson
Dept. of Psych., Atkinson College, York Univ., ON, Canada

11:00–11:10 Discussion

11:10

3aSC4. Music and speech in a neuropsychological perspective. Isabelle Peretz (Dept. of Psychol., Univ. of Montreal, CP 6128, Centre-ville, Montreal, PQ H3C 3J7, Canada)

Neuropsychology — defined as the study of the relations between brain organization and mental functioning — has concerned itself from its earliest days with the processing of speech and music. The first steps were taken in the speech domain, via the observation of selective speech impairments following damage to specific areas of the brain (Broca, in Von Bonin, *Some Papers on the Cerebral Cortex*; Wernicke, in Brain Function **3**, 1–16). These initial findings swiftly prompted the exploration of music disorders (Bouillaud, Bull. Acad. Med. **30**, 752–768). Such observations are labeled aphasia and amusia, respectively, and reflect a long-standing concern to describe the selectivity of the disorders observed. This issue of specificity is even more prevailing/acute in contemporary neuropsychology, with the advances made in brain imaging and in experimental psychology. Current knowledge will be summarized and organized along two main questions: (1) Is music, like speech, subserved by neutral circuitries devoted to its processing? (2) What are the boundaries of neuropsychological separability between music and speech?

11:35–11:50

Commentary by Kenneth Pugh
Haskins Labs. and Yale Medical School, New Haven, CT

11:50–12:00 Discussion

12:00–12:15 Open Discussion

Session 3aUW

Underwater Acoustics: Propagation

Andrew A. Piacsek, Chair
Lawrence Livermore National Laboratories, L-200, P.O. Box 808, Livermore, California 94550

Chair's Introduction—8:15

Contributed Papers

8:20

3aUW1. Reproducibility of low-frequency shallow-water acoustic experimental data. Richard B. Evans (SAIC, 21 Montauk Ave., Ste. 201, New London, CT 06320) and W. M. Carey (ARPA, Arlington, VA 22203)

Transmission loss measurements were conducted in September 1993 on the continental shelf off New Jersey. Low-frequency continous-wave data were recorded on a vertical line receiving array in approximately 73 m of water. The geometry of these measurements replicates the geometry of experimental measurements conducted during October 1988 at the same site and under similar downward refracting conditions. The sets of frequencies transmitted during the two experiments were different, but covered the region less than 1 kHz. Acoustic surveys and geophysical data obtained in 1988 provided detailed characterizations of the seabed. The bottom characterization is assumed to be time invariant, except for the contribution of the pore fluid to the uppermost layers of the sediment. Consequently, the characterization of the seabed should be valid when the bottom waters are similar. The coincidence of the two sets of experimental data and a valid geoacoustic model provide the opportunity to test the reproducibility of these results. The bottom characterizations based on two sets of acoustic data are compared with expectations based on profiles derived from geophysical data.

8:35

3aUW2. Recent shallow-water acoustic measurements of sound transmission, reverberation, and coherence. William M. Carey (Adv. Res. Projects Agency, 3701 N. Fairfax Dr., Arlington, VA 22203), Peter Cable, and Mike Steele (BBN Systems and Technol., Arlington, VA 22209)

Recent experiments performed in the shallow waters (100 m deep) of the Gulf of Mexico, East Continental Shelf of America, and the Korean Straits under the condition of downward refraction with sandy–silty clays are discussed. These experiments were performed with both explosive, air gun, and continuous sources covering the 100-Hz to 1-kHz frequency range. The receivers were both horizontal arrays on the bottom and vertical arrays which spanned the water column. Precise navigation was employed to eliminate range uncertainties and all measurement systems were calibrated to a traceable standard. The results are presented in the categories of sound transmission, reverberation, and coherence. Sound transmission results were found to generally agree with wave-theoretic numerical codes and shear was not found to be important. The reverberation was found to have a frequency-dependent characteristic consistent with sediment layering. Coherence lengths are estimated from array signal gain measurements in the midfrequency range and compared to previous measurements in both deep and shallow water. Results were found to be influenced by a variety of factors such as water column variations. Estimates of the importance of these factors under downward refracting conditions will be made and extended to other sound velocity profiles.

8:50

3aUW3. Interference patterns of frequency-dependent transmission loss for shallow-water propagation. Kevin P. Bongiovanni, Mohsen Badiey (Graduate College of Marine Studies, Univ. of Delaware, Newark, DE 19716), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Transmission loss (TL) as a function of frequency, for fixed source and receiver positions, provides a means to investigate the influence of environmental mechanisms on broadband acoustic propagation. These curves exhibit an analogous behavior to cw loss curves as a function of range, with null-peak interference patterns observed in both experimental data and numerical simulation. An analytic approximation for the resulting patterns in a Pekeris type range-independent medium employing normal-mode theory is developed. Broadband signals are partitioned into bands for which accurate modeling can be achieved. Narrow-band approximations are then used to derive formulas for the null-spacings of the curves. These expressions connect the interference structure, through the modal group velocities and appropriate boundary conditions, to volume and seabed structure and composition. This work extends a previous development for a zeroth-order ocean [Badiey *et al.*, J. Acoust. Soc. Am. **98**, 2897(A) (1995)]. A qualitative assessment of the formulation is achieved by comparisons with both parabolic equation simulations and high-resolution geoacoustic/acoustic experimental data, for several different environments. The evolution of TL versus frequency patterns in range will be discussed in light of range-dependent environmental parameters.

9:05

3aUW4. Statistical analysis of broadband propagation in shallow water. Mohsen Badiey, Simon A. Shaw, Kevin P. Bongiovanni, Thomas C. Honsinger, and Joe R. Zagar (Ocean Acoust. Lab., Graduate College of Marine Studies, Univ. of Delaware, Newark, DE 19716)

One of the most important issues facing ocean acoustic experimentalists when attempting to perform calibrated experiments is distinguishing measurement errors from environmental variabilities. These errors include accuracy of monitoring hydrophones, ship positioning (hanging sources), and, for a broadband source in particular, spectral level and phase variations of the source. Statistical analysis is used to investigate the repeatability of source and receiver signatures collected from a series of broadband experiments performed in very shallow water at the Atlantic Generating Station (AGS) site. Classification of the factors inducing signal variation, which may arise from surface and bottom roughness and time-varying volume inhomogeneities, in addition to source errors, is made for several experimental configurations. It is shown that the coherence of the received signals is directly related to the environmental effects on the acoustic waves and the shot-to-shot correlation decreases with increasing range.

9:20

3aUW5. Frequency dependence of forward propagation through a layered dipping sediment. Roger M. Oba (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

Forward propagation is used to compute the acoustic field in a shallow-water environment with a layered dipping sediment. The sediment is assumed to be silty with several thin sandy layers with higher density and sound speed. The sandy layers slope within the entire sediment layer so that the depth of a sand layer varies linearly with range. The sediment is modeled as a fluid of uniform horizontal thickness over a range-independent hard sub-bottom. The water sound speed is range independent and depth dependent in a fixed depth environment. This range-dependent sediment geometry leads to transmission loss significantly different from those with fixed layer depths, even when the dip gradient is very slight. These differences and their mechanisms are explored for varying frequencies and interlayer distances.

9:35

3aUW6. A study of the influence of an off-shore rise on low-frequency modal propagation with Arctic surface loss. Thomas N. Lawrence, Wade Trappe, and Nancy R. Bedford (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713)

Simple up-slope propagation of underwater acoustic energy can usually be modeled by an adiabatic normal mode approximation, provided the slope is not too severe. Bottom interacting paths in such an environment are usually sufficiently attenuated to make consideration of mode coupling unnecessary. An environment further complicated by an off-shore rise can cause additional acoustic energy to be introduced into lower-order modes due to bottom interaction. These low-order modes will then propagate in deep water until encountering the continental shelf. Experimental results from the Arctic Ocean between 25 and 45 Hz suggest such propagation conditions. These results, and their interpretation in the context of a coupled mode study, will be presented. Environmental parameters and source depth will be varied to explore under what conditions mode coupling can be ignored when considering detection and localization problems. Arctic surface loss will be applied to the modeling results to show how mode varying attenuation will affect those results. The coupled mode model (COUPLE) [R. B. Evans, J. Acoust. Soc. Am. **74**, 188–195 (1983)], is used as the vehicle for this study. [This work is sponsored by SPAWAR PMW 182-2.]

9:50

3aUW7. Improved narrow-band and broadband normal-mode algorithms for fluid ocean environments. Evan K. Westwood (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Improvements have been achieved in the speed, robustness, and versatility of a recently developed normal-mode algorithm for fluid ocean environments [S. J. Levinson *et al.*, J. Acoust. Soc. Am. **97**, 1576–1585 (1995)]. The possibility of missing modes has been eliminated by computing both the total phase of the oscillatory mode function versus depth and the number of zeros of the function. The root-finding algorithm computes both an error function and its derivative, which allows cubic interpolation to be used to obtain eigenvalue guesses when a root is bracketed. The efficiency of the root-finder is characterized by the fact that the error function typically must be computed less than 3-1/2 times per mode. An automated broadband capability has also been implemented. Analytic first and second derivatives are used to interpolate the eigenvalues in frequency as a fifth-order polynomial. For broadband computations where inclusion of the continuum is desired, a false bottom may be automatically inserted. Its thickness is specified in terms of acoustic wavelengths in the medium and varies with the frequency, thus saving the computation of a significant number of modes. [Work supported by ONR 321OA.]

10:05–10:20 Break

10:20

3aUW8. Acoustic propagation through a low Mach number, stratified flow. Pierre Elisseeff and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Propagation of sound in an oceanic waveguide in the presence of a low Mach number, stratified flow is investigated. Equations for the wavenumber integration (WI) and normal mode (NM) approaches are simultaneously derived in a single formulation. When hydrodynamic shear stress and acoustic azimuthal coupling are negligible, the effect of current can be accounted for by a simple modification of existing WI and NM codes. A current layer is then shown to act as a wave-number filter which, in some cases, cannot be modeled using an equivalent sound-speed profile. Broadband numerical simulations show very good agreement between independently modified versions of OASES and KRAKEN. Applications to tomography are eventually discussed. The acoustic field for smooth current profiles can be derived as a function of the medium-at-rest mode set, making a broadband inversion possible. [Work supported by ONR.]

10:35

3aUW9. Invariant imbedding analysis of seabed acoustics. Khaldoun Khashanah (Dept. of Mathematical Sciences, Stevens Inst. of Technol., Hoboken, NJ 07030) and Thomas G. McKee, Jr. (Stevens Inst. of Technol., Hoboken, NJ 07030)

The linear acoustic equations for the bounded slab of fluid ocean and elastic seabed with cylinderical symmetry are reduced to a set of coupled boundary value problems using separation of variables. In order to have a computational technique capable of handling stratification effects in the seabed, invariant imbedding is used to replace the boundary value problem with an initial value problem. As an initial approximation, the case of an ocean with a reflective bottom is considered and the change in the solution, as the depth of the seabed increases, is calculated. The method of invariant imbedding is shown to be numerically stable and has the advantage of assessing the effects of including an interactive seabed on the solution to the underwater acoustics problem with a reflecting seabed.

10:50

3aUW10. Mode coupling analysis for a resonance effect due to an internal soliton. Michael K. Broadhead and Robert L. Field (Code 7173, Naval Res. Lab., Stennis Space Center, MS 39529-5004)

As pointed out by Zhou *et al.* [J. Acoust. Soc. Am. **90**, 2042–2054 (1991)], nonlinear shallow-water solitary internal waves (SIW's) can enhance the bottom interaction of underwater sound. For a lossy ocean bottom, this has the effect of a level change (in addition to fluctuations) in the transmission loss at preferred frequencies. The mechanism for this effect is acoustic mode coupling due to the depression of higher sound-speed water into lower speed water (at the pycnocline). It is also possible for this mechanism to induce a transfer of acoustic energy from below the thermocline into the mixed layer, as has been shown elsewhere, where a two-layer pycnocline/sound-speed environment was used, along with a KdV soliton. Over the frequency range studied (900–1030 Hz), the energy transfer spectrum exhibited a doublet resonant structure. It was also found that an increase in the SIW length scale was accompanied by a positive shift in the resonant frequencies. A simple model is offered that qualitatively accounts for these two features which is based on mode coupling theory. Also presented is the dependency of the resonance frequencies on the sound speeds and layer thicknesses of the acoustic environment.

11:05

3aUW11. Nonlinear wide-angle paraxial propagation in shallow-water channels. Rahul S. Kulkarni, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180-3590), and Michael D. Collins (Naval Res. Lab., Code 7140, Washington, DC 20375-5320)

A model that describes wide-angle paraxial propagation of acoustic pulses in shallow water has been developed. This time-domain, range-marching model incorporates weak nonlinear effects and depth variability in both ambient density and sound speed, with extensions to include dissipative effects. The derivation is based on an iterative approach, in which the wide-angle approximation is obtained by using a narrow-angle equation to approximate the second range derivative in the two-way nonlinear wave equation. Scaling arguments are used to obtain a more tractable simplification of the equation. Higher-order approximations can be derived by continuing the iterative procedure. The wide-angle equation is solved numerically by splitting it into components representing distinct physical processes and employing a predictor-corrector strategy. A high-order upwind flux-correction method is used to handle the nonlinear component, in order to eliminate spurious artifacts that otherwise degrade the solution. Numerical results are presented for different types of sources in several horizontally stratified environments. Effects of nonlinearities on wide-angle propagation and differences between narrow- and wide-angle nonlinear propagation will be discussed. [Work supported by ONR.]

11:20

3aUW12. Estimation of nonlinear effects in global scale ocean acoustic propagation. B. Edward McDonald (Naval Res. Lab., Washington, DC 20375)

The nonlinear progressive wave equation (NPE) [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)] is used to derive quantitative estimates for the magnitudes and characteristic features of nonlinearities generated by finite amplitude acoustic signals during propagation across ocean paths of global scale. Nonlinear effects accumulate over long propagation paths, and are accentuated at convergence zone caustics. The goals are to estimate (1) levels of harmonic generation in ocean tomographic experiments and (2) waveform steepening in underwater explosions. Harmonic generation (1) is estimated from a linear baseline solution involving vertical eigenmode expansion. The result implies second harmonic magnification at odd numbered convergence zone ranges. Waveform steepening (2) is estimated by time-domain numerical integration and related to modal excitation differences in the far field [J. Ambrosiano, D. R. Plante, B. E. McDonald, and W. A. Kuperman, J. Acoust. Soc. Am. **87**, 1473–1481 (1990)]. [Work supported by ONR.]

11:35

3aUW13. Propagation of signals from strong explosions above and below the ocean surface. Douglas B. Clarke, Andrew A. Piacsek, and John W. White (Lawrence Livermore Natl. Lab., L-200, P.O. Box 808, Livermore, CA 94550)

In support of the Comprehensive Test Ban, research is underway on the long-range propagation of signals from nuclear explosions in the deep underwater sound (SOFAR) channel. Our work has emphasized the variation of wave properties and source region energy coupling as a function of height or depth of burst. Initial calculations on CALE, a two-dimensional hydrodynamics code developed at LLNL by Robert Tipton, were linked at a few hundred milliseconds to a version of NRL's weak shock code, NPE, which solves the nonlinear progressive wave equation [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)]. The wave propagation simulation was then followed down to 5000-m depth and out to 10 000-m range. In the future, calculations on a linear acoustics code will extend the propagation to greater distances. Until recently our research has considered only explosions in or above the deep ocean. New results on energy coupling and signal propagation in shallow water and the effects of other improvements will be presented. [Work performed under the auspices of the U. S. Department of Energy by the Lawrence Livermore National Laboratory under Contract W-7405-ENG-48.]

WEDNESDAY MORNING, 15 MAY 1996 EXECUTIVE BOARDROOM, 9:00 A.M. TO 12:00 NOON

Meeting of Accredited Standards Committee S2 on Mechanical Vibration and Shock

to be held jointly with the

U. S. Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock

D. J. Evans, Chair S2

National Institute of Standards and Technology, Acoustics, Mass and Vibrations Group, Building 233, Room A147, Gaithersburg, Maryland 20899

D. F. Muster, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 108, Mechanical Vibration and Shock

4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees), including a report on the activities of ISO/TC 108, with the plans for its September 1996 meeting in Sydney, Australia.

Scope of S2. Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical vibration and shock, and condition monitoring and diagnostics of machines, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Session 3pAA

Architectural Acoustics and Musical Acoustics: Directivity of Musical Instruments II

Uwe J. Hansen, Chair
Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Chair's Introduction—2:00

Invited Paper

2:05

3pAA1. The influence of directivity on sound heard by an audience during an orchestral performance. Jürgen Meyer (Phys.–Tech. Bundesanstalt, Bundesallee 100, D-38116 Braunschweig, Germany)

For a listener seated in a hall, sound impressions are created by the time structure and the intensity ratios of direct sound, early reflections, and reverberation. The intensity ratios between these components depend on the directivity of the sound generated by different instruments. Even for high-pitched instruments, omnidirectional radiation is found only at frequencies below 500 Hz. The strength of the directivity can be described by the statistical directivity factor which fixes the angle-dependent reverberation radius of a source sounding in a hall. For directions of main radiation at upper string frequencies it is of the order of 2–3; for the woodwinds, it is a little lower; for the brasses it exceeds 4. Directions of strong radiation oriented toward the sidewalls, thus supporting the space impression, are found with the strings, bassoons, horns, and trombones. High-frequency components of strings are radiated in strong measure against the ceiling: These reflections support the brilliancy. These sound effects will be demonstrated with a real orchestra.

WEDNESDAY AFTERNOON, 15 MAY 1996 REGENCY BALLROOM, 12:45 TO 1:50 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Mauro Pierucci, Chair
Department of Aerospace and Engineering Mechanics, San Diego State University, San Diego, California 92182-1311

Chair's Introduction—12:45

Invited Papers

12:50

3pID1. A hot topic in musical acoustics: Phase coherent representation of radiated sound fields. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809)

A technique used to study normal-mode vibrations in musical instruments is modal analysis. This experimental method analyzes a series of transfer functions which relate the frequency spectra of a number of impact points to the spectrum of a fixed response point, and subsequently presents the motion of the structure at the various normal-mode frequencies in computer animation. This approach is extended to represent a radiated sound field by using a fixed excitation point on the musical instrument as the reference signal and monitoring the sound field with a microphone at a number of predetermined points. This makes it possible to relate the phase of the sound field to the phase of the vibrating structure responsible for generating that field. Animated illustrations will be shown for the air column in a wind instrument and for the sound field of a concert grand piano.

1:10

3pID2. Hot topics in engineering acoustics. Steven Garrett (Graduate Program in Acoust. and Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

Engineering acoustics attempts to utilize contemporary developments in many areas of science, technology, and analysis to produce acoustical systems and products which address existing needs or create new opportunities. Topics in engineering acoustics become "hot" when there is a convergence which facilitates a "quantum leap" in some particular application area. Fusion of new sensor technologies, microprocessors, signal processing, and electronic data communication has produced new options in condition-based monitoring of machinery and industrial processes. Advances in the speed and reduction of cost in A–D and D–A converters, coupled with high memory density, has permitted the development of acoustic time reversal mirrors which could have application to ultrasonic nondestructive evaluation of materials. Computing power, new inversion algorithms, and novel transducers have led to the commercialization of resonant ultrasound spectroscopy which can determine the complete set of elastic moduli for materials with sample sizes which can be less than 1 mg. The availability of fast, accurate one-dimensional computer-based acoustic modeling tools and new research results in studies of the transfer to heat from acoustically oscillating fluids to solid surfaces are providing a new foundation to design and optimize the next generation of thermoacoustic engines and refrigerators.

1:30

3pID3. The study of individual differences in the auditory system. Dennis McFadden (Dept. of Psych., Mezes Hall 330, Univ. of Texas, Austin, TX 78712)

Individual people often produce reliably different results on various auditory tests even when all have nominally normal hearing. An emerging topic in hearing research is the origins of these individual differences in normal-hearing people. Examples of procedures used to measure the potential contribution of genetics and hormone exposure to individual differences will be discussed, and several examples of a special instance of individual differences in hearing–sex differences–will be described. Of considerable value for the ultimate understanding of individual differences are situations in which subsets of people differ in a similar way across different auditory measures because covariations of this sort have the potential to reveal common underlying mechanisms.

WEDNESDAY AFTERNOON, 15 MAY 1996 MT. RUSHMORE, 1:00 TO 3:15 P.M.

Session 3pPA

Physical Acoustics and Bioresponse to Vibration and to Ultrasound: Workshop on Therapeutic Applications of Medical Ultrasound IV: Tour to Indiana University School of Medicine

Robert E. Apfel, Chair
Yale University, New Haven, Connecticut 06520-8286

Registration for this tour will be conducted at session 2pPAa on Tuesday afternoon in Regency C. Space is limited.

Invited Paper

1:00

3pPA1. Shock wave lithotripsy: A demonstration of experimental methods for *in vitro* shock wave exposure and analysis of cell injury. James A. McAteer, Sharon P. Andreoli, Andrew P. Evan, Dax D. Denman, Coleen Mallett (Depts. Anatomy and Pediatrics, Indiana Univ. Med. School, Indianapolis, IN 46202), Robin O. Cleveland, Michalakis A. Averkiou, Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105), James E. Lingeman, and David Lifshitz (Methodist Hospital of Indiana, Indianapolis, IN 46202)

This laboratory is involved in studies to determine the mechanism(s) responsible for cellular injury to the kidney during shock wave lithotripsy for the treatment of upper urinary tract stones. The studies are performed using an electrohydraulic lithotripter (Dornier HM3) and injury to isolated, cultured renal tubular epithelial cells was assessed. In this demonstration our experiences with the practical aspects of performing *in vitro* SW exposures are shared. Issues to be discussed and/or demonstrated include the following: (a) choice of cultured cell models and the handling of isolated cells, (b) factors that influence *in vitro* cell injury, (c) selection and preparation of specimen vials, (d) lithotripter setup and operation, (e) positioning the specimen and control of exposure parameters, (f) electrode performance and variability, (g) characterization of waveforms and determination of SW pressures, (h) markers of cell injury and cell response to SW insult, and (i) safety in the lithotripsy suite. [Work supported by NIH Program Project Grant No. PO1 43881: Role of SWL in renal injury and stone comminution.]

3p WED. PM

Session 3pSA

Structural Acoustics and Vibration: Fuzzy Structures

Geoffrey L. Main, Chair
Office of Naval Research, Code 334, 800 North Quincy Street, Arlington, Virginia 22217

Contributed Papers

12:30

3pSA1. Numerical solutions of a prototypical master structure/fuzzy substructure system. Richard L. Weaver (Dept. of Theoretical and Appl. Mechanics, 216 Talbot Lab., Univ. of Illinois, 104 S. Wright St., Urbana, IL 61801)

The transient response of a single degree of freedom master oscillator attached to a simple undamped N degree of freedom "fuzzy substructure" is studied numerically and theoretically. Results at early times are found to be in accord with the predictions of the Pierce–Sparrow–Russell theory; in particular, the master oscillation manifests an apparent damping. At later times, however, the energy is returned from fuzzy to master. The precise manner in which the energy is returned and the time taken to do this depend on the details of the mass and frequency distribution within the fuzzy and, in particular, on the distribution of spacings between the fuzzy resonances. For the case of irregularly positioned fuzzy resonances the energy returns immediately and the master then oscillates randomly. For the case of regularly spaced fuzzy resonances the energy returns after a longer time, and does so coherently. Theory is presented which supports the accuracy of the Pierce–Sparrow–Russell result at short times. Other arguments (for the case of random fuzzy resonances) predict the root-mean-square level of the subsequent random oscillations. Still others (for the case of regularly spaced fuzzy resonances) predict the return time. [Work supported by ONR.]

12:45

3pSA2. Equipment representations as time-domain fuzzy structures. Aravind Cherukuri and Paul E. Barbone (Dept. of Aerospace & Mech. Eng., Boston Univ., 110 Cummington Ave., Boston, MA 02215)

The presence of complex subsystems can dramatically affect the dynamics of the main structure to which they are attached. Exactly modeling these subsystems, however, is often impossible. Here, replacing the substructure (say a piece of equipment) by an equivalent set of forces which react back on the main structure is proposed. These forces are given as time convolutions of the displacements at the equipment attachment points. The convolution integral, which represents a time-domain DtN (Dirichlet-to-Neumann) map, is approximated in the high modal density limit with determined error bounds. Local in time approximations to the convolution integral are obtained using Padé approximants. These yield a family of equipment representations. The simplest requires two measured equipment properties, though more information can lead to greater accuracy. Our approximate DtNs are validated numerically in finite-element simulations. [Work supported by ONR.]

1:00

3pSA3. Experimental identification of fuzzy structure parameters. Raymond J. Nagem, Ann W. Stokes, and Allan D. Pierce (Dept. of Aerospace and Mech. Eng., Boston, Univ., 110 Cummington St., Boston, MA 02215)

Researchers have recently been developing a theory to model structures in which a primary or "master" structure with precisely known parameters is coupled to "fuzzy" substructures with parameters known only in a statistical sense. The goal of the fuzzy theory is to model complex structures with as few parameters as possible. The research presented here is an experimental investigation of the effectiveness of fuzzy parameter choices in modeling the drive-point impedance of a beam with a number of simple oscillators attached along its length. The fuzzy model makes use of "most likely" distributions of mass per unit resonance frequency, given limited knowledge about the attached oscillators. The oscillators are interchanged in order to achieve various realizations of mass per unit natural frequency. The magnitude and phase of the measured drive point impedance are compared to that predicted by fuzzy theory models. The use of the moments of the mass per unit natural frequency distribution as the primary descriptors of the effect of the fuzzy substructures is examined. Bounds on the differences between predicted and measured drive point impedances are established. The use of these bounds in applications such as high-performance, robust vibration control is discussed.

1:15

3pSA4. Vibration response statistics in regular and irregular plates. Christopher A. Hartemink and Richard G. DeJong (Eng. Dept., Calvin College, Grand Rapids, MI 49546)

Statistical measures of the vibration response of flat plates in both the frequency and spatial domains are presented. Analytical, numerical, and experimental results are presented for rectangular and irregularly shaped plates. The results are consistent with the assumption that the modal response of a rectangular plate is sinusoidal, while the modal response of an irregularly shaped plate is Gaussian. The statistical measures of a plate vibration response can be used to evaluate the system modal overlap factor, the ratio of the modal damping bandwidth to the modal frequency spacing.

1:30

3pSA5. Development of a mid-frequency definition and appropriate analysis techniques. John E. Huff, Jr. (School of Mech. Eng., Purdue Univ., 1077 Herrick Labs., West Lafayette, IN 47907-1077), Guglielmo Rabbiolo, Robert J. Bernhard, and Fabio A. Milner (Purdue Univ., West Lafayette, IN 47907)

From experimental data collected on ensembles and from analytical verification studies of structural acoustic systems, it is apparent that there is a mid-frequency range where dominant structural acoustic responses cannot be sufficiently described deterministically or statistically. The focus of this presentation is to describe work to derive a general and rigorous definition for the frequency limits of the mid-frequency range. The statistical behavior of system response in this range is described through the analysis of the analytical expressions for the displacement of Bernoulli–Euler beams. Additionally, work to develop a numerical analysis technique which can be implemented in a finite element formulation is discussed and the possible connections between it and current techniques for energy flow analysis in the high-frequency range are made.

1:45

3pSA6. Forced localization in mistuned nonlinear repetitive structures. Melvin E. King (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

The presence of localized modes in repetitive structures (i.e., systems composed of identical substructural elements) often gives rise to motions during which vibrational energy becomes spatially confined to a subset of elements. Such modes have been shown to be generated through eignevalue veering in weakly mistuned linear systems and mode bifurcations in perfectly tuned nonlinear systems. Recent work by the author has investigated the combined influences of weak nonlinearities *and* weak structural mistunings in generating localized modes. In the present work, the forced response of nonlinear cyclic systems with structural mistunings is investigated via the method of multiple scales. Under harmonic excitations, strongly and weakly localized motions will be shown to exist for various structural parameters. Sample calculations will be presented for systems composed of two, three, and four degrees of freedom. Additionally, motion confinement characteristics of such systems will be demonstrated for transient loading conditions, and the implications for novel vibration isolation designs will be discussed.

WEDNESDAY AFTERNOON, 15 MAY 1996 CIRCLE THEATER, 3:15 TO 5:15 P.M.

Plenary Session, Business Meeting, and Awards Ceremony

Robert E. Apfel, Chair

President, Acoustical Society of America

Business Meeting

Vote on changes to ASA Certificate of Incorporation

Announcement of Gold and Silver Certificate Recipients

Presentation of Certificates to New Fellows

Presentation of Awards

Gold Medal to Ira Dyer

R. Bruce Lindsay Award to Victor W. Sparrow

Presentation of President's Tuning Fork

Presidential Address
"Acoustics in 2016"

NOTE: The Town Meeting will immediately follow the Plenary Session at the Circle Theater.

Session 4aAA

Architectural Acoustics: Design and Measurement in Architectural Acoustics

Dana S. Hougland, Chair
David L. Adams Associates, Inc., 1701 Boulder Street, Denver, Colorado 80211

Chair's Introduction—9:00

Contributed Papers

9:10

4aAA1. A musical history of architectural acoustics. Christopher Herr and Gary W. Siebein (Architecture Technol. Res. Ctr., 231 ARCH, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611-5702)

A brief multimedia presentation explaining the mutual developments of acoustics, music, and architecture was developed to illustrate the relations among art, science, culture, and music in the evolution of western architecture. Throughout much of the history of the western world there has been a clearly discernible connection between the development of music, musical instruments, ensembles, and the design of spaces for musical performances. This multimedia presentation was developed to illustrate the creative potential of understanding music and acoustics in the history of architecture. The acoustical rationales for musical and architectural phenomena are studied in the context of composers and musical works. The musical pieces have been convolved with the acoustical properties of the spaces in which they were originally performed to allow people to aurally experience a simulated acoustical history of architecture. Examples studied include chants in large cathedrals; Gabrieli's music composed for St. Mark's; comparisons of Haydn's works written early in his career for small orchestras in small rooms and those written later in his career for larger orchestras in larger rooms; the operatic works of Wagner and the unique environment at Bayreuth; and the exploration of tone color and experimental music and architecture in the 20th century among others [Work supported by NSF.]

9:25

4aAA2. The duality of the observer and the observed—From quantum mechanics to architectural acoustics. Robert T. Beyer (Dept. of Phys., Brown Univ., P.O. Box 1843, Providence, RI 01912)

In 1928, Niels Bohr pointed out the dual description required in the quantum mechanics of measurement, involving the observer and the observed [Naturwissenschaften **17**, 483–486 (1928)], and a few years later, John von Neumann discussed this principle at some length in his text on the *Mathematical Foundations of Quantum Mechanics* [(Springer-Verlag, Berlin, 1932), English translation, Princeton U.P., 1955]. It now appears that architectural acoustics has been following the same route. Examples cited by von Neumann are described and paralleled with examples from the work of Schroeder, Ando, and others in concert hall acoustics.

9:40

4aAA3. Comparisons of room acoustics measurement systems. J. S. Bradley (Acoust. Lab., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

This paper gives a summary of the results of an international room acoustics measurement system round robin. Twenty-three different measurement systems were compared by measuring three settings of the same digital reverberator that was shipped to each measurement group. This was equivalent to testing three standard rooms, but the tests did not include variations due to different types and placements of transducers. The various measurements were made over a one year period and the reverberator and measurements were shown to be stable and very reproducible over this period. The means of the closely clustering results were used to represent the best estimate of the "correct" answers. The standard deviations of the closely clustering results were used to evaluate the divergence of the measurements from the central trend. Approximately $\frac{2}{3}$ of the results were within two of these standard deviations from the central trend. Two specific problems were identified and solutions recommended. While most results were in close agreement, a number of results differed significantly from the central trend and demonstrated the importance of validating new room acoustics measurement systems.

9:55

4aAA4. Automated architectural acoustical modeling with the SeDReS system. Gary W. Siebein, Mitchel E. Spolan, and Christopher Herr (Architecture Technol. Res. Ctr., 231 ARCH, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611-5702)

A computer-controlled architectural modeling system was developed to allow the enclosure and materials of a room to be rapidly and precisely varied for acoustical testing using ultrasonic sparks and miniature microphones at various scales (depending upon the size of the model room to be tested). A system of stepper motors moves segmented wall, ceiling, and floor panels of a large, flexible enclosure. The walls, ceiling, and floor can be moved to configure a wide variety of rooms from a small lecture room to a large arena. The locations and angles of the enclosing surfaces of the room for multiple design motifs can be stored in the memory of the computer. One can then push several keystrokes to reconfigure the room and proceed with acoustical testing and simulation using the acoustical research instrumentation for architectural spaces (ARIAS) system. Impulse responses, acoustical measurements, and aural simulations from alternative designs can be compared at the earliest stages of the design process when one is considering basic choices of finish materials, room shape, and wall angles. [Work supported by NSF.]

10:10

4aAA5. Analysis and synthesis of room reverberation in the time and frequency domains—Application to the restoration of room impulse responses corrupted by measurement noise. Jean-Marc Jot, Guillaume Vandernoot, and Olivier Warusfel (IRCAM, 1 Pl. Stravinsky, 75004 Paris, France)

This paper reviews a statistical time-frequency model of the late diffuse reverberation in rooms. In this model, the later part of a room impulse response is described by a "time-frequency envelope," defined as the ensemble average of the time-frequency energy distribution in the late decay. The time-frequency envelope is independent of the source and receiver locations and is characterized by two functions of frequency: the reverberation time and a power spectral density called the "initial spectrum," which corresponds to the product of the diffuse-field transfer func-

tions of the source and the receiver. An analysis method is described for deriving these parameters from a measured impulse response, yielding an accurate estimate of the reverberation time with arbitrary resolution in the frequency domain. The time-frequency envelope defines a time-dependent filter which transforms a sampled stationary Gaussian white noise into a synthetic model of the late reverberation decay. Substituting this synthetic reverberation decay to the part of an impulse response that is corrupted by measured noise, yields a restored impulse response allowing faithful auralization by convolution with anechoic source signals. The models and the methods described are illustrated on impulse responses measured in a concert hall.

10:25

4aAA6. Analysis of subjective acoustic measures and speech intelligibility in Portuguese churches. António P. O. Carvalho, António Morgado (Acoust. Lab., Dept. of Civil Eng., College of Eng., Univ. of Porto, R. Bragas, 4099 Porto Codex, Portugal), and Luís Henrique (School of Music-ESMAE, 4000 Porto, Portugal)

This study reports on subjective acoustical field measurements made in a survey of 36 Catholic churches in Portugal built in the last 14 centuries. The same group of college students were asked to judge the quality of speech and music at all the churches. Two sets of listeners in each church evaluated live music performance (cello and oboe) at two similar locations in each of the rooms using a seven-point semantic differential rating scale. An acoustical evaluation sheet was used to measure listeners overall impression of room acoustics qualities, and each of the factors that can contribute to that perception as *loudness, reverberance, intimacy, envelopment, balance, clarity, echoes*, and *background noise*. Speech intelligibility tests were also given to the same group in each church. One-hundred-word lists were used in live speech tests using a theater college student as speaker. The results are graphed and analyzed by comparisons. Variations of subjective and speech intelligibility qualities were identified among the different churches and within each of the churches as well. For instance, church mean values for the intelligibility scores range from 56% to 96% correct words and the variability within each room extent from 6% to 49%. The subjective qualities that contributed to overall acoustical impression were also identified. [Work supported by CEDEC, ESMAE and UP-Portugal.]

10:40

4aAA7. A proposed acoustical renovation for The Great Hall at The Cooper Union. Gregory A. Miller (Dept. of Mech. Eng., The Cooper Union, 51 Astor Pl., New York, NY 10003) and Daniel Raichel (The Cooper Union and CUNY, New York, NY 10003)

The Great Hall is an 1100-seat auditorium with historical landmark status built in 1859. It has been the site of many historic speeches by figures ranging from President Abraham Lincoln to Sioux Chief Red Cloud. Throughout its history, however, The Great Hall has been described as a less than adequate acoustical environment, suffering from low speech intelligibility and excessive reverberance. Building upon the previous work of Raichel and Dragan who modeled the auditorium in its current configuration, the authors undertook a study of the feasibility of renovating and improving The Great Hall while maintaining the architectonic integrity of the space, as necessitated by historical landmark protection. It was found that a simple replastering of the walls and ceiling of the auditorium will lower the reverberation time at 250 Hz from over 3 s to under 1 s and should significantly reduce the effect of late (after 50 ms) reflections. The new plaster will have lower density and higher porosity than the old plaster, significantly improving its sound-absorbing characteristics. The proposed renovations are currently under consideration for implementation.

10:55

4aAA8. Comparative study of modular and standard construction methods for broadcast/recording studio applications. David Michalek and Brandon Tinianow (Acoustic Systems, P.O. Box 3610, Austin, TX 78764)

Consultants and owners are becoming increasingly aware of the choice between modular versus conventional construction methods when designing or remodeling broadcast and recording studios. In a laboratory environment the physical attributes such as acoustic isolation, direct material costs, physical weight, etc. have been quantified and the resulting cost per dB of attenuation determined. When assessing the method to be employed, the less obvious yet vital differences often overlooked include the consultants' added liability associated with conventional construction methods (risk of failure of acoustical isolation), supervision required during construction to insure attention to critical sound isolation details, durability, installation time, shortage of qualified workmen, fit and finish, exposure to unexpected on-site delays, ability to retrofit, or relocate, etc. These design and cost considerations will be compared and contrasted.

11:10

4aAA9. The effects of special coatings on the Sabine–Guastavino Rumford acoustical tiles. Richard Pounds (Graduate School of Architecture, Planning and Preservation, Columbia Univ., New York, NY 10027), Daniel R. Raichel (Cooper Union, New York, NY 10003 and CUNY), and Martin Weaver (Columbia Univ., New York, NY 10027)

Wallace C. Sabine and Rafael Guastavino developed the Rumford tiles specifically to lessen the reverberation times inside ecclesiastical edifices. These tiles are used in a number of churches, including St. Bartholomew and St. Thomas, both in New York City, and the Duke University Chapel in Durham, NC. In recent times, those responsible for musical programs have been demanding greater reverberation times, even at the expense of clarity of spoken words. Accordingly sealers are being applied to a number of church interiors in order to increase the reverberation times. Because of the limited samples available, the tone burst method was applied at the Cooper Union Acoustics Research Center to measure the effect of applying special coatings to decrease sound absorption coefficients. The absorption coefficients were measured for the naked tiles, provided by the First Congregational Church of Montclair, NJ, and after the consolidator was sprayed on. The test was also repeated for each application of the sealer. The results of the test indicate that a considerable amount of coating will be necessary to decrease the sound absorption coefficient to the proper degree.

Session 4aAB

Animal Bioacoustics: Intra-Specific Variations in Animal Vocalizations I

Robert H. Benson, Chair
Center for Bioacoustics, MS 3367, Texas A&M University, College Station, Texas 77843-3367

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Song displays, song dialects, and lek mating systems in hummingbirds. S. L. L. Gaunt (Borror Lab. Bioacoustics, Dept. Zool., Ohio State Univ., Columbus, OH 43210)

Males of some hummingbirds, including Colibri (violet-ears), congregate and display from traditional areas, leks, to attract mates. Hummingbirds use a vocal display. Males on a lek share the same song, i.e., nearest neighbors share song; males on distant, acoustically isolated, leks have different songs, i.e., there are dialects. Dialects have been used as evidence for song learning in birds, and, although song learning is usually ascribed to oscines and parrots, hummingbirds share vocal organ (syrinx) structures with them consistent with song learning ability. Digital spectrogram cross correlation was used to objectively determine degree of song similarity; this statistic was used in cluster analysis to describe relationships between birds and with the Mantel method to test hypotheses about associations. Further, males of a lek deliver their song cooperatively, i.e., without interference. The resultant is a lek signal that, from a distance, sounds as one song. If, as in other cooperative lekking systems, there is a skew in benefits, i.e., few obtain mates, then other males may gain benefits indirectly through relatedness (kin selection). Results from preliminary, DNA "fingerprinting" do not support this hypothesis, and alternative explanations are suggested. [Work supported, in part, by NSF.]

8:35

4aAB2. Recognition of the utterances of terrestrial wildlife: A new approach. Ronald P. Larkin (Illinois Natural History Survey, 607 E. Peabody Dr., Champaign, IL 61820), Daniel Margoliash, Joseph A. Kogan (Univ. of Chicago), and Larry L. Pater (US Army CERL)

Finding and censusing birds and other animals via listening can pose problems because of inaccessibility of habitats, rarity or shyness of animals, or subjectivity of observers. A new collaborative project seeks to evaluate algorithms adapted from human speech recognition to establish a basis for automating the identification of animal vocalizations and recording their occurrence. The algorithms include dynamic time warping and hybrid hidden Markov models incorporating features of artificial neural networks. Probably no single method will work for all species. More than one method maybe useful together, in multiple stages. A database of high-quality, annotated digital field recordings is being collected to supply training and test data on known species and, when possible, known individuals. Both low-noise and realistic ambient noise situations are important. Field data are supplemented with recordings from laboratory settings. Red-cockaded woodpecker, other vocal yet threatened species, and species related to them, such as other Picoides woodpeckers, are being studied. Preliminary results are presented. [Research supported by USACERL.]

9:05

4aAB3. Intraspecific variation in primate vocalizations. Jeffrey C. Norris (Ctr. for Bioacoustics, Texas A&M Univ., 5007 Ave. U, Galveston, TX 77551)

Intraspecific variability in primate vocalizations exists at several levels—within organizational levels of a species, as well as differingly across acoustic variables. This paper reviews signal variability between subspecies, groups and populations, individuals, and within a single individual's vocalizations. The problem of signal variability is best seen when humans describe a species' repertoire. The literature is replete with examples where humans underestimate the amount of variability recognized by the animals; humans typically are lumpers, where playback experiments show that the animals recognize differences that are imperceptible to us. The perceptual cues used by five primate species will have been described. Sensitivity to variability is not uniform between acoustic variables. Variability in signal production has been described for an additional 14 species. Details of interindividual and intraindividual variability of wedge-capped capuchin alarm calls is described. Where early taxonomies of this species' vocalizations recognized two alarm calls, subsequent research by the author described 15 variants of one call, at least three of which were recognized by the monkeys to refer to different objects. The importance of the vocalization's function will be stressed, recognizing that motivational calls may have greater intrinsic variability than those referring to specific external objects.

9:35–9:50 Break

9:50

4aAB4. Sperm whale codas in the northwestern Gulf of Mexico. Troy D. Sparks, William E. Evans (Marine Acoust. Lab., Ctr. for Bioacoustics, Texas A&M Univ., 5007 Ave. U, Galveston, TX 77551), and Robert H. Benson (Texas A&M Univ., College Station, TX 77843-3367)

Sperm whale (*Physeter macrocephalus*) coda patterns from the northwestern Gulf of Mexico have been examined. Recordings were collected via a towed passive hydrophone array. Sperm whales are known to produce a characteristic "hammering" click. Codas are rhythmic patterns of clicks [W. A. Watkins and W. E. Schevill, J. Acoust. Soc. Am. **62**, 1485–1490 (1977)]. Two types of codas have been proposed: identity codas and general use or shared codas [Watkins *et al.*, Cetology **49**, 1–15 (1985)]. Identity codas are patterns that are unique to an individual for at least a few hours, and general use codas are vocalizations that seem to be shared among groups of whales. In the southeast Caribbean, it was found that 50% of the codas analyzed were made up of two patterns; therefore, it was proposed that shared codas may have a function other than individual identity [Moore *et al.*, Mar. Mam. Sci. **9**, 1–9 (1993)]. There also appears to be geographical differences in codas' spacing and composition between the Galapagos and the southeast Caribbean [L. Weilgart and H. Whitehead, Can. J. Zoo. **71**, 774–752 (1992)].

10:05

4aAB5. A comparison of the songs of the tufted and black-crested titmice. Cade L. Coldren (Dept. of Wildlife and Fisheries Sciences, Texas A&M Univ., College Station, TX 77832-2258)

The songs of the Eastern tufted titmouse (*Parus B. bicolor*) and the black-crested titmouse (*Parus B. sennetti*) were examined as a possible systematic tool. From nine sites, 65 song types were identified, falling into four categories: one-note phrase, two-note long-short phrase, two-note short-long phrase, and three-note phrase songs. Canonical discriminant analysis on morphometric parameters revealed greater variation in song between sites than within sites. Black-crested titmice typically sing at higher dominant frequencies, shorter phrase durations, a higher number of phrases per song, and a smaller repertoire than tufted titmice. Black-crested repertoires were composed mostly of one-note phrase songs and a few two-note phrase songs. Tufted titmice repertoires had approximately equal proportions of one-note and two-note phrase songs with one three-note phrase song at each site. Differences in body size and habitat density may explain the higher dominant frequencies found at black-crested sites. The possible occurrence of dialects and the role of learning may account for all differences in song. However, their impacts on titmice songs must be determined before song can be used as a reliable systematic tool for the tufted titmouse complex.

10:20

4aAB6. Geographic variation and cultural evolution in songs of humpback whales (*Megaptera novaeangliae*) in the eastern North Pacific. Salvatore Cerchio (Museum of Zoology, Univ. of Michigan, 1109 Geddes Ave., Ann Arbor, MI 48109) and Jeff Jacobsen (P.O. Box 4492, Arcata, CA 95521)

Songs of humpback whales off Hawaii and Mexico were examined to determine whether they changed similarly in both areas during a breeding season. Songs of 24 individuals were recorded off Kauai, Hawaii and Isla Socorro, Mexico during winter and spring of 1991. Forty-seven acoustic variables describing all levels of song structure were measured for each singer. Similar variables were grouped together into six categories. Mean values for each singer were compared among regions and time periods using two-factor ANOVAs. All but three variables changed between winter and spring in at least one area. Groups of similar variables displayed similar trends. Quantitative characteristics of song elements often changed during the breeding season by the same amount in each area, with little variation within and among individuals. Structures of song patterns often changed differently in each area. Results indicated cultural transmission may have occurred during the season. Alternatively, whales may be predisposed to gradually change features of song independent of cultural influences. Given the low level of variability exhibited by many variables, the latter seems more likely; therefore, temporal change, like song structure, may be governed by a discrete set of rules. Further research on a longer time scale is needed.

10:35

4aAB7. The use of passive sonar to detect sound production and calculate population densities of penaeid shrimp in the Gulf of Mexico. Ilona M. Berk, William E. Evans (Ctr. for Bioacoustics, Marine Acoust. Lab., Texas A&M Univ., 5007 Ave. U, Galveston, TX 77551), Robert H. Benson, and Michael E. Duncan (Texas A&M Univ., College Station, TX)

Past research concluded that nonstridulating pink shrimp (*Penaeus duorarum*) found in the Gulf of Mexico did not make sound in the sonic range, but might produce ultrasonic sound via friction between parts of the exoskeleton [W. R. Gehring, Sea Grant Bulletin #5 (1971)]. Recordings of "frying" noises over known penaeid shrimp beds led to speculation that the shrimp were mechanically producing the "frying" portion of the ambient noise, and that passive sonar might be used as a tool for penaeid shrimp detection and estimation of population densities [W. E. Evans, Acoustics Signature Catalog, 82–141 (1982)]. Sound production by nonstridulating white shrimp (*Penaeus setiferus*) of the Gulf of Mexico was studied using passive sonar. Video and hydrophone recordings were made of captive shrimp populations with known densities. Hydrophone recordings of wild populations were made pre- and postshrimp trawls, and trawl catch data were noted. Results showed that the captive shrimp produced detectable mechanical sound primarily via escape movements and possibly mastication. Initial comparison suggests that wild shrimp and captive shrimp sounds are similar. It is hoped that the final results of analysis will confirm that wild penaeid shrimp population density calculations are feasible using passive sonar data.

10:50

4aAB8. The acoustics of snapping shrimp in Kaneohe Bay. Whitlow W. L. Au (Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734) and Kiara Banks (Florida Inst. of Technol., Melbourne, FL 32901)

Snapping shrimp are among the major contributors of biological noise in shallow bays, harbors, and inlets located in temperate and tropical waters. Snapping shrimp sounds can severely limit the use of underwater acoustics by humans and other animals such as dolphins, whales, and pinnepeds. They produce sounds by rapidly closing their frontal chela, or claws, snapping the two ends together to generate a loud click. The acoustics of the species *Synalpheus paraneomeris* was studied by measuring the sound produced by individual shrimp housed in a small cage located 1 m from an H-52 broadband hydrophone. Ten clicks from 40 specimens were digitized at a 1-MHz sample rate and the data stored on disk. Various acoustic parameters such as source level, spectral content, and center frequency were correlated with claw size and body length. Peak-top-peak source levels varied from 183 to 189 dB *re*: 1 μPa. A typical spectrum had a low-frequency peak between 2 and 5 kHz and energy extending out to 200 kHz.

11:05

4aAB9. Intraspecific variation in the vocalizations of the *willet Catoptrophorus semipalmatus*. Hector D. Douglas (Dept. of Biology, Wake Forest Univ., P.O. Box 7325, Reynolda St., Winston-Salem, NC 27106)

The acoustical repertoire of the eastern willet was recorded at sites along the Atlantic seaboard using a Marantz portable cassette recording deck (model PMD 221), a Sennheiser microphone (model K3-U), and a 62-cm fiberglass parabolic recording dish. Associated behaviors were annotated on audio tape and in field notebooks. Call parameters were measured using a DSP Sona-Graph model 5500. Variation in call parameters was assessed within individuals, within populations, between geographical regions, and between subspecies using cross correlations and univariate and multivariate statistical methods. Sonograms, preliminary quantitative results and aural comparisons suggest relatively little structural variation in calls across large geographical regions. One notable exception in this pattern was observed at the subspecific level, and a hypothesis is advanced linking differences in ecology and environmental features to explain this divergence. Some intraindividual variation could be linked to hypothesized transitions in motivational states as indicated by incidental changes in behavior. Results suggest that vocal development in the willet is based upon innate templates. However, observations also suggest that these forms may be recombined to encode different information.

11:20

4aAB10. Geographic and seasonal variation in blue and finback whale vocalizations. David K. Mellinger (Bioacoustics Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850) and Christopher W. Clark (Cornell Univ., Ithaca, NY 14850)

The U.S. Navy maintains a network of listening arrays for detection of submarines. For several years, some of the sounds from these arrays have been made available to scientists for study of cetacean movements and vocalizations. Here the vocal behavior of blue (*Balaenoptera musculus*) and finback (*B. physalus*) whales are analyzed. These species produce long sequences of calls, lasting hours to days, that exhibit several types variation across areas of the North Atlantic and Pacific. Finback produce two commonly heard general types of calls, with other types present less frequently. Finbacks in different geographic areas exhibit variations in frequency, in intercall timing, in the mixture of call types, in patterns of calls and silences, and in other ways. Blue whales, unlike finbacks, show distinctly different vocalizations in the Atlantic and Pacific; but like finbacks, both Atlantic and Pacific whales have two principal call types, with different mixtures of the types in different areas. Differences in timing and frequency are heard in distinct areas, with certain patterns heard only in one or a few areas. [Work supported by ONR.]

11:35

4aAB11. Objective analysis of song learning in birds: Toward automated techniques. S. E. Anderson, C. A. Staicer, S. Inoue, and D. Margoliash (Dept. of Organismal Biology and Anatomy, Univ. of Chicago, 1025 E. 57th St., Chicago, IL 60637)

It is demonstrated that indigo buntings sing bouts of "adult plastic" songs (APS) distinct from stereotyped songs (SS). Some distinguishing characteristics of APS include syllables of variable morphology and syllables not present in SS. The hypothesis that APS may play a role in adult song learning is tested by monitoring socially paired males housed together. Some yearlings changed their SS to more closely match their tutors' SS, by replacing syllables or inserting new syllables. New syllables were developed in APS through transformation and combination of morphologically similar existing syllables, and syllables were transferred from APS into SS. To address the limitations of manual scoring of spectrographs, dynamic time warping (DTW) for template-based automated analysis of continuous recordings of birdsongs was evaluated. With laboratory recordings, the DTW algorithm employed identified syllables and syllable boundaries of SS and calls of indigo bunting and zebra finch with greater than 97% accuracy. APS constituents were identified with approximately 84% accuracy. Under these restricted recording conditions, DTW has general applicability to objective analysis of birdsongs, and dramatically decreases the effort required. Application of hybrid hidden Markov models may improve the performance for variable vocalizations such as APS, and under noisy conditions of field recordings.

THURSDAY MORNING, 16 MAY 1996 NATIONAL PARKS, 8:00 TO 11:45 A.M.

Session 4aEA

Engineering Acoustics: General Engineering Acoustics

Alison B. Flatau, Cochair
Aerospace Engineering and Engineering Mechanics, Iowa State University, 2019 Black Engineering Building, Ames, Iowa 50011

Frederick T. Calkins, Cochair
Aerospace Engineering and Engineering Mechanics, Iowa State University, 2019 Black Engineering Building, Ames, Iowa 50011

Contributed Papers

8:00

4aEA1. Mode control of ultrasonic guided waves in thick cylinders for crack detection. Zongbao Li and Yves H. Berthelot (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Experiments on ultrasonic propagation in thick annular structures show that one can simplify the modal structure of the received signals by carefully controlling the wedge angles of the generation and detection transducers. This is important if one is to detect cracks along the path between source and receiver. In particular, one should excite and detect only those modes whose energy is concentrated where the cracks are expected to form. To do so, the following methodology is used: First, experimental signals are obtained for various source–receiver configurations with a standard pulser/receiver; a wavelet transform is then applied to the signals to localize precisely in time the arrival of a given mode packet at a given frequency and to determine the corresponding group velocity. These modes are then identified by comparing the results with predictions from theoretical dispersion curves. One can then predict the energy distribution within the structure by using the method of normal mode expansion for transients. [Work supported by the ONR, Code 332.]

8:15

4aEA2. Practical implications of micromachined ultrasonic transducers. Igal Ladabaum and Butrus T. Khuri-Yakub (Stanford Univ., E. L. Ginzton Lab., Stanford, CA 94305)

Micromachined ultrasonic transducers capable of airborne transmission of 1- to 12-MHz ultrasound have been reported. It has also been reported that these transducers should enable ultrasonic systems with 100 dB of dynamic range. A justification of the predicted dynamic range involves the careful consideration of the thermal noise limits of the system. Such analysis is developed and is based on the fluctuation dissipation theorem. In addition, the first scans produced with the new transducers are presented along with interpretations of their significant impact to nondestructive testing. Pictures of surface and bulk defects in composites and metals are included. Finally, the fabrication process is summarized and the issues involved in the development of immersion transducers are highlighted. [Work supported by the U.S. Office of Naval Research.]

8:30

4aEA3. Modification of two methods for determining acoustic impedance. Michael G. Jones (Lockheed Martin Eng. and Sciences Co., 144 Res. Dr., Hampton, VA 23666) and Patricia E. Stiede (NASA Langley Res. Ctr., Hampton, VA 23681)

Results of an investigation to modify two impedance measurement methods (two-microphone method, TMM; and multipoint method, MPM) to improve efficiency and convenience are presented. The TMM is modified by eliminating the need for one of the microphones in situations where only one microphone can be placed into the test environment. The MPM modification uses pseudorandom noise to significantly reduce the time necessary to complete acoustic impedance measurements. These modifications are the results of on-going technology development at NASA Langley Research Center to simplify and increase accuracy of measurements in harsh environments. Evaluations of efficiency and accuracy requirements of each of the normal incidence acoustic impedance measurement techniques in use at NASA Langley Research Center are provided to determine which method should be used for a given problem. The single-tone source implementation of the MPM is found to be the most accurate, but the most time-consuming technique currently in use. A pseudorandom noise source implementation of the MPM requires significantly less time, with minimal loss of accuracy. The single-microphone method (SMM) is shown to be accurate with a single discrete frequency source, but degrades in accuracy with a random noise source.

8:45

4aEA4. Acoustic radiation by a rattling plate: Experimental and theoretical analysis. Karen J. Lee and Karl Grosh (Dept. of Mech. Eng. and Appl. Mechanics, Univ. of Michigan, Ann Arbor, MI 48109-2125)

Rattle noise is directly tied to customers' perceptions of product quality. The accurate and efficient prediction of this noise component remains an important and challenging area. Sound-pressure levels (SPLs) in the acoustic field of a rattling plate were measured experimentally and compared to predictions based on the plate's equations of motion. The aluminum plate was hinged at one end and contacted a point at the other. Rattle was induced by exciting the plate supports using a shaker with a sinusoidal input. SPL measurements were made with a microphone at different points in the acoustic field. For the initial theoretical predictions, the plate was modeled as a rigid body and a closed-form solution was found for the linearized equations of motion to obtain the contact forces. The calculated force was used to generate the predicted plate vibration which, in turn, was used to predict the SPL in the acoustic field surrounding the plate. Later in the study, plate flexibility was added into the contact force model, and the effects of adding an increasing number of vibrational modes were studied. The experimental and theoretical data were compared in the time and frequency domains.

9:00

4aEA5. Advances in acoustic pyrometry. John A. Kleppe (Elec. Eng. Dept., Univ. of Nevada, Reno, NV 89557-0153)

Instrumentation based on acoustic principles is finding widespread use in utility boiler applications. This paper describes how acoustic systems are currently being used to measure volumetric flow rate in large ducts and stacks for continuous emission monitoring system (CEMS) applications as well as for the measurement of high-temperature flue gases and boiler control. Acoustic pyrometry is a useful tool for the measurement of gas temperatures in the furnace and superheater regions in boilers. Results have shown that acoustic pyrometry is easy to apply, accurate, and noninvasive. Gas temperature measurements are provided and displayed on a continuous real-time basis for single path volumes in various furnace regions and for multiple path arrays in the furnace exit gas plane. The determination and presentation of isothermal maps of gas temperature provide a new tool for operation and maintenance diagnostics, not achievable by any other currently known method. Acoustic pyrometers are now being used to help identify and correct burner problems, slagging problems, and furnace overheating before these conditions can adversely affect operations.

9:15

4aEA6. Comparisons of approximate and exact techniques for convected acoustics. Laurine Leep (Comput. Aided Eng. Dept., Ford Res. Lab., Dearborn, MI 48121) and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

In many aero- and environmental-acoustic problems, convection of sound waves is handled by appropriately increasing or decreasing the local speed of sound and then solving the resulting Helmholtz equation. Such solutions are typically obtained via integral transforms or geometrical acoustics. An exact solution technique that properly incorporates the vector character of mean-flow convection has been found [L. Nijs and C. P. A. Wapenaar, J. Acoust. Soc. Am. **87**, 1987–1998 (1987)]. Unfortunately, the formal and numerical complexity of these techniques has prevented detailed comparisons of the acoustic field variables with and without the standard approximation. Results and comparisons for approximate and exact treatment of convection will be presented from a series of simple computations made directly from the linearized time- and space-dependent equations of inviscid fluid motion. Specifically, the propagation of initially plane acoustic waves in a simple shear flow has been addressed to determine how: (i) wavefront orientation and location, (ii) acoustic-particle-velocity vector direction, and (iii) acoustic wave strength compare when mean flow convection is handled approximately and exactly. [Work supported by Ford Motor Company.]

9:30

4aEA7. Time capture method for improvements in absolute calibration of accelerometers by laser interferometry. Lixue Wu (Associated Standards, Inst. For Natl. Measurement Standards, Natl. Res. Council Canada, Ottawa, ON K1A 0R6, Canada)

The measurement uncertainty in the absolute calibration of accelerometers using laser interferometry was reduced by a time capture method. Outputs of a photodetector and accelerometer were digitally sampled at a rate of 256k samples per second. Displacement amplitudes and accelerometer sensitivities were found from the sampled data. During data processing, a technique was used that decomposes a multivariable nonlinear optimization into a series of single-variable optimizations. This decomposition technique not only reduces the computational complexity but also avoids the convergence to local minima. Results of calibrations showed that the measurement uncertainty at the high-frequency end (800 Hz) for a small acceleration of 10 ms^{-2} was less than 0.9% at 95% confidence level. This time capture method can be applied to any time-variant (nonsinusoidal, nonperiodic) displacement measurement. With the above method, measurement requirements such as frequency stability and acceleration amplitude stability, can be less stringent. The effects of airborne noise and vibration can also be reduced.

9:45–10:00 Break

10:00

4aEA8. A conical shock tube that simulates the heavyweight shock tests in MIL-S-901D. L. D. Luker and A. L. Van Buren (Underwater Sound Reference Detachment, Naval Undersea Warfare Ctr. Div. Newport, P.O. Box 568337, Orlando, FL 32856-8337)

Sonar transducers and related components mounted on the hulls of ships and submarines are subject to both inertial shock and acoustic pressure shock when a weapon explodes in the water near the vessel. Consequently, sonar transducers are required to pass an open-water explosive shock test using the floating shock platform (FSP) as specified in MIL-S-901D(NAVY). These tests are expensive, infrequent, and susceptible to erroneous results due to damage incurred in shipment of the transducers between the shock test site and the acoustic calibration facility. This paper describes the development of a conical shock tube (CST) facility at NUWC-USRD that is an inexpensive, rapid-turnaround alternative to open-water testing. CST tests are performed indoors in a closed, water-filled tube with a 25-cm-diam muzzle opening. The CST matches the peak pressures observed in the FSP test using only several grams of explosive. It also matches the axial motion experienced by the FSP in open-water tests. Both design criteria and experimental results are presented. [Work supported by NAVSEA.]

10:15

4aEA9. Dynamic shear modulus measurement technique for very soft materials. Dimitri M. Donskoy and Gyoo-Cheol Bang (Davidson Lab., Stevens Inst. of Technol., Hoboken, NJ 07030)

Existing methods of dynamic viscoelastic properties measurements have some limitation with respect to softness of material, its spatial anisotropy, and frequency range. Specifically, there are very limited capabilities to measure dynamic shear moduli of very soft (real shear modulus value is below 10^4 Pa) and anisotropic materials such as biological tissue and some polymers. The proposed technique is intended to fill this niche. The method utilizes the resonance approach, in which the specimen with certain orientation is treated as a massless spring attached to a massive rigid plate. The plate is excited with a one-dimensional forced vibration in such a way that the specimen has only shear deformation. The resonance oscillation of the plate is analyzed with an impedance head and a spectral analyzer. Once the resonance frequency and the quality factor of the resonance are obtained, the complex shear modulus of the specimen can be calculated. An experimental setup utilizing the proposed approach was built and the complex shear moduli of soft biological tissue with respect to different orientations of the fibers were measured.

10:30

4aEA10. Sonar dome deflection and strain measurement system. Joel F. Covey (Naval Res. Lab., Code 7135, Washington, DC 20375-5350), Diane B. Weaver, and Jonathan B. Walker (SFA, Inc.)

A multiplexed computer-controlled deflection and strain measurement system has been developed for use in performing at-sea data collection on a Naval surface ship sonar dome rubber window. The deflection of the window is measured at 128 points on its water-filled inner surface by an acoustic time-of-flight technique. At each point of interest a 2-cm-diam hemispherical acoustic transducer is pulsed at 110 kHz and the signal is received by an array of four similar transducers fixed to the ship's sonar array. Ten different receiver arrays are used to allow for direct line-of-sight reception for each of the points of interest. The 85 strain sensors were built into the wire plies of the window by the manufacturer (B. F. Goodrich). The system records data from all 128 transducers, all strain sensors, as well as numerous environmental parameters at a rate of approximately 200 reading/s. The measurement system is nonobtrusive to normal ship sonar operations and performance. The data collected is being used in characterizing the behavior of these windows at high speeds and sea states and in the calibration of numerical experiments and window redesign efforts. [Work supported by NAVSEA.]

10:45

4aEA11. Measured in-water acoustic performance of 1-3 piezocomposite materials. Thomas R. Howarth (Naval Res. Lab., Code 7130, Washington, DC 20375-5350)

A selection of different 1-3 piezocomposite materials has been fabricated and tested. Each of the 1-3 piezocomposite materials that will be presented was fabricated by the Material Systems Inc. of Littleton, MA. The materials selection includes variations in volume content of piezoceramic, types of piezoceramic, host matrix materials of varying compliances and air-voided microballoon content, as well as the use of types of cover plates on the radiating surfaces. The acoustic measurements were conducted at both the Lake Gem Mary lake facility and the Anechoic Test Facility 1 (ATF1) of the Underwater Sound Reference Detachment in Orlando, FL. The presentation will include comparisons of free-field voltage sensitivities (FFVS) and transmitting voltage response (TVR) as well as directivity patterns. The measurements from ATF1 will also show the effects of hydrostatic pressure and temperatures of 4 ° and 22 °C. The presentation will include a discussion of how specific 1-3 piezocomposite materials can be tailored to meet specific applications. [Work supported by Advanced Research Projects Agency and NSWC/Coastal Systems Station.]

11:00

4aEA12. Experimental evidence for maximum efficiency operation of a magnetostrictive transducer. Frederick T. Calkins and Alison B. Flatau (Dept. of Aerospace Eng. and Eng. Mechanics, Iowa State Univ., Ames, IA 50011)

Experimental broadband testing of a magnetostrictive transducer has been used to study the trade-off between operating for maximum efficiency and maximum output. Although the largest output, displacement or acceleration, is achieved at the resonant frequency, a greater operating efficiency is achieved at a frequency above resonance. An explanation for this phenomenon for general transducers via the theory of electroacoustics is reviewed [F. V. Hunt, *Electroacoustics: The Analysis of Transduction, and Its Historical Background*, Acoustical Society of America, Woodbury, NY (1982)]. This frequency of maximum efficiency is derived for a general electromagnetic transducer and shown graphically on a motional-impedance locus plot. An evaluation of the efficiency shows that the reactance, the imaginary component of the electrical impedance, can be 'tuned' to achieve maximum efficiency. For a magnetostrictive transducer the reactance tuning is related to reducing eddy losses. The effective permeability of the material is shown to decrease significantly after resonance, well below the nominal value at low frequency; thus operating in this frequency region results in a significant decrease in eddy current losses and hence a higher efficiency. Experimental results are provided to demonstrate the advantages of operating a transducer at the maximum efficiency frequency. [Work supported by NASA Graduate Student Research Program, NASA Langley.]

11:15

4aEA13. Analysis of *tonpilz* with flexing pistons. Alan H. Lubell (Lubell Labs., Inc., 21 N. Stanwood Rd., Columbus, OH 43209) and Ralph Simon (1777 Westwood Ave., Columbus, OH 43212)

The acoustic loading of each of the first six modes of a double piston piezoelectric stack-driven underwater acoustic transducer is found using Botman's method [NAA report NA65H-1024, June 1965] revised with the use of the Helmholtz integral formulation of Chertok [J. Acoust. Soc. Am. **36**, 1305–1313 (1964)]. Both self and mutual impedances are obtained and the admittance of each mode solved for. Finite element analysis is used to determine the mode parameters of the first 30 modes and the contribution to compliance loading of modes 7-30. Using symmetry, the eight-element stack is represented by a ten-mesh circuit that includes joint compliances, end piece masses, a distributed parameter representation of the stressbar, and the contact compliances between end pieces and pistons. Input impedance and total radiated power are computed for a wide range of frequencies. For joint compliance of the order of 0.25 E−10 M/N or less, a simpler stack model is found to be adequate.

11:30

4aEA14. Relation between total power output and input phase of underwater transducer array elements made of piezo-rubber. Daiji Mikami, Akio Hasegawa, and Toshiaki Kikuchi (Dept. of Appl. Phys., Natl. Defense Acad. of Japan, 1-10-20 Hashirimizu, Yokosuka 239, Japan)

The mutual interaction of underwater acoustic transducer array elements is one of the most important factors for its transmission performance. The power output from each array element is characterized by mutual radiation impedance between array elements. Recently numerical simulations are often applied to solve this kind of problem. Generally PZT transducer array is used in experiments but mechanical Q of PZT transducer is high and the range of measuring frequency is limited in narrow band. So many sizes of transducer array models are required to evaluate in different frequencies. Composite material piezo-rubber has unique properties. Its flexibility and broadband resonance character by low mechanical Q allow experiments in various frequencies utilizing one array apparatus. Consequently a flexible method of model experiment using piezo-rubber sheet transducer array is introduced. The total power output versus input phase of 3×3 piezo-rubber transducer array provides the effects of interaction between array elements in various frequencies and input amplitudes. Also near field 3-D mapping of real power and imaginal power shows radiation performance of this transducer array visually. A clear power reduction was observed in the case of a -180-deg phase shift pattern.

THURSDAY MORNING, 16 MAY 1996 HARRISON'S, 9:40 A.M. TO 1:00 P.M.

Session 4aED

Education in Acoustics: Demonstrations in Acoustics

Mardi C. Hastings, Cochair
Department of Mechanical Engineering, The Ohio State University, Columbus, Ohio 43210-1107

Janet M. Weisenberger, Cochair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, Columbus, Ohio 43210

Chair's Introduction—9:40

Invited Paper

9:45–11:45

4aED1. Lecture demonstrations in acoustics. Fredericka Bell-Berti (St. John's Univ., Jamaica, NY 11439 and Haskins Labs., New Haven, CT 06511), E. Carr Everbach (Swarthmore College, Swarthmore, PA 19081-1397), Mardi C. Hastings (Ohio State Univ., Columbus, OH 43210), Patricia K. Kuhl (Univ. of Washington, Seattle, WA), and Janet M. Weisenberger (Ohio State Univ., Columbus, OH 43210)

A series of lecture demonstrations in acoustics will be given to high-school students from Indianapolis area high schools. Attendance and participation by meeting attendees are welcome.

Session 4aNS

Noise: Progress Report and Discussion on the Continuing Activity of ASA's Role in Noise and its Control

Louis C. Sutherland, Chair
27803 Longhill Drive, Rancho Palos Verdes, California 90275

Invited Paper

11:00

4aNS1. Noise: Progress report and discussion on the continuing activity of ASA's role in noise and its control. Bennett M. Brooks (Brooks Acoust. Corp., Vernon, CT 06066), T. James DuBois (Acentech, Canoga Park, CA 91303), Robert M. Hoover (Hoover and Keith, Inc., Houston, TX 77082), George C. Maling (Empire State Software, Ltd., Poughkeepsie, NY 12603), and Louis C. Sutherland (Rancho Palos Verdes, CA 90275)

A discussion meeting sponsored by TC Noise is being held to review progress on activity underway or proposed. Highlighted will be a summary of an all-day seminar on plant and product noise control, led by Dr. Luc Mongeau, to be presented Friday, May 17 to industrial engineers, designers and planners invited to this outreach effort by ASA. The report will also include an update on hearing testing activity, presentations on noise and acoustics to school classes by task group and other ASA members and the status of a joint proposal by ASA and INCE submitted to the Society of Automotive Engineers to add a noise element to their continuing award-winning series of "World in Motion" educational packages to promote science and mathematics literacy for grades 6–8. The report will also include discussion on inadequate acoustic environments in school classrooms and the resulting potential for negative impact on the learning process for children. In discussions with Stanley L. Ehrlich, president-elect of ASA and Dana S. Hougland, chair of TC Architectural Acoustics, Robin M. Towne proposed this topic as worthy of ASA involvement. Thus members of TCAA and other interested persons are also encouraged to attend this session.

THURSDAY MORNING, 16 MAY 1996 REGENCY A, 8:00 TO 11:00 A.M.

Session 4aPA

Physical Acoustics: Nonlinear

Bart Lipkens, Chair
Macrosonix Corporation, 1570 East Parham Road, Glen Allen, Virginia 23130

Contributed Papers

8:00

4aPA1. Nonlinear surface wave propagation in crystals. M. F. Hamilton, Yu. A. Il'inskii, and E. A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

Elastic wave propagation along the free surface of a crystal differs from Rayleigh wave propagation in an isotropic solid. In particular, the phase speed and depth dependence vary with the direction of propagation with respect to the crystallographic axes, and the wave-vector component perpendicular to the surface is complex. Results from a theoretical investigation of nonlinear propagation along a free surface of a crystal are presented. The analysis is based on spectral equations that are derived with Hamiltonian formalism. The equations were integrated numerically to illustrate finite-amplitude distortion of a surface wave that is sinusoidal at the source. In one example, calculations based on published material properties are presented for surface wave propagation in KCl out to distances beyond where shock formation occurs. Waveform distortion associated with propagation along the [111] axis in KCl was found to be similar to that of Rayleigh waves in isotropic solids. However, for propagation along the [001] axis in KCl, energy transfer from the fundamental to the second harmonic component is very inefficient, and the waveform distortion was found to differ considerably from that of Rayleigh waves. [Work supported by NSF, ONR, and the Schlumberger Foundation.]

8:15

4aPA2. Second harmonic generation in a sound beam incident on a liquid–solid interface near the Rayleigh angle. B. J. Landsberger, M. F. Hamilton, Yu. A. Il'inskii, and E. A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

When a beam of sound is incident on an elastic half-space at the Rayleigh angle, the rapid variation of the phase of the reflection coefficient produces the classical effect known as beam displacement. Considered here is the corresponding effect on the second harmonic component generated nonlinearly in the fluid. Experiments were performed in water with a 1-MHz beam radiated from a circular source of radius 1.2 cm. The beam was incident on an aluminum block at several angles near and at the Rayleigh angle. A quasilinear theory was developed in which the primary beam is decomposed into its angular spectrum. Analytic solutions were derived for second harmonic generation by pairs of angular spectrum components in the incident and reflected fields. The analytic solutions are superposed, and the second harmonic field is constructed from the resulting angular spectrum. There are no restrictions on source geometry or angle of incidence. Theory is in excellent agreement with experiment. Calculations reveal that in the reflected field, second harmonic components generated separately in the incident and reflected beams combine construc-

tively in regions associated with specular reflection, and destructively in regions associated with radiation by the Rayleigh wave. [Work supported by ONR and the Schlumberger Foundation.]

8:30

4aPA3. Nonlinear acoustic nondestructive testing of cracks. Alexander M. Sutin (Inst. of Appl. Phys., Russian Acad. Sci., 46 Ulyanov str., Nizhny Novgorod 603600, Russia)

This paper is a review of new nonlinear acoustic methods. Acoustic imaging and NDT methods are widely used in various tecnical fields. Conventional methods are based on the principles of linear acoustics: Nonlinear distortions are ignored. The strength of the distortion can highly increase due to the compliant features of the cracks. Such distortion can be observed by using different nonlinear effects such as high-harmonic generation, modulation of sound by vibrations, and subharmonic generation. Nonlinear acoustic methods are based on these effects and examples of its usage are varied. First, a similar method was developed which tested the glue quality of thermoprotective covers of the Russian Buran space shuttle. Using a nonlinear effect of high-frequency sound modulation by vibrations, an enormous increase of the acoustic nonlinear parameter in steel that has been fatigued has been experimentally demonstrated. The single microcracks in metal construction were detected as well. The method based on second-harmonic generation due to cracks was tested in firm "PECHINEY" in France. The sensitivity of nonlinear new methods to the appearance of fatigue damage to the material is much larger than that of conventional methods of NDT. [Work was supported by Russian Foundation of Fundamental Research Grant N0-93-05-8074.]

8:45

4aPA4. Propagation of finite-amplitude broadband noise. Penelope Menounou and David T. Blackstock (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029 and Mech. Eng. Dept., Univ. of Texas, Austin, TX 78712-1063)

Burgers' equation is used to predict the effect of nonlinearity on the power spectral density of plane broadband noise traveling in a nondispersive thermoviscous fluid. The source signal is assumed to be stationary Gaussian noise, which, because of nonlinear propagation distortion, becomes non-Gaussian as it travels. As opposed to time-domain methods, the method presented here is based directly on the power spectral density of the signal, not the signal itself. The Burgers equation is transformed into an unclosed set of linear equations that describe the evolution of the joint moments of the signal. A method for solving the system of equations is presented. Only the evolution of appropriately selected joint moments needs to be calculated in order to predict the evolution of the power spectral density of the signal. The results are in good agreement with a time-domain code [Cleveland *et al.*, J. Acoust. Soc. Am. **98**, 2865(A) (1995)]. The method can be also applied when the source condition is a stationary, ergodic, and Gaussian stochastic process. [Work supported by NASA.]

9:00

4aPA5. Absorption of sound by noise in one dimension. Andrés Larraza (Dept. of Phys.—Code PH/La, Naval Postgrad. School, Monterey, CA 93943) and Bruce Denardo (Univ. of Mississippi, University, MS 38677)

Experimental results are presented for the excess attenuation of monofrequency waves in the presence of high-intensity noise in one dimension. The theory by Rudenko and Chirkin predicts a Gaussian attenuation of a weak signal in the presence of shockless noise, thus showing that in one dimension translational invariance has been broken. The theory is modified to incorporate wall losses providing, only then, excellent agreement with experiment. The agreement is shown as the frequency, noise level, and distance from the source are varied. In addition, the spectral intensity of the high-frequency tail of fully developed shockless noise is observed to be an f^{-3} power law in the frequency f, in accord with theory. This power law is a consequence of the far off equilibrium nature of the system. [Work supported by ONR.]

9:15

4aPA6. Nonlinearity versus diffraction within a focusing weak shock. Andrew A. Piacsek (Lawrence Livermore Natl. Lab., Univ. of California, P.O. Box 808, Livermore, CA 94551)

In a classic paper treating the propagation of weak shocks, Whitham elucidated the process by which nonlinear effects would prevent rays within a concave portion of a shock front from intersecting, thereby avoiding a folded wavefront and associated caustics [G. B. Whitham, J. Fluid Mech. **1**, 290–318 (1956)]. Whitham's analysis neglected the effects of diffraction arising from a finite shock thickness. When this length scale is small compared to that of the wavefront's curvature, Whitham's nonlinear correction to geometrical acoustics is a good model for the development of the shock front. When the length scales are comparable, as they are near the focal point, diffraction becomes important and acts both in conjunction with nonlinearity (to limit the shock amplitude) as well as against it (inhibiting the refraction of rays). Recent numerical studies [A. Piacsek, Ph.D. thesis, The Pennsylvania State University (1995)] indicate that the latter effect can lead to the formation of a folded wavefront, bounded by caustics, just as linear geometric theory predicts. The present paper addresses quantitatively how the relative importance of nonlinearity and diffraction (as represented by the shock amplitude, thickness, and wavefront curvature) determines the behavior of a focusing weak shock. [Work performed by the Lawrence Livermore National Laboratory under U. S. Department of Energy Contract No. W-7405-ENG-48.]

9:30–9:45 Break

9:45

4aPA7. Generation of localized low-frequency vibrations by remote ultrasound sources. Mostafa Fatemi and James F. Greenleaf (Biodynamics Res. Unit, Dept. of Physiol. and Biophys., Mayo Clinic/Foundation, Rochester, MN 55905)

This paper presents a new method of inducing localized low-frequency acoustic energy in an object by remote sources. In this method two interacting high-power, high-frequency, ultrasound beams at two different frequencies are used. These beams are positioned to interact at the object's surface (or any other impedance discontinuity inside the object). The result is a radiation force at the difference frequency, df, exerted in a small area defined by the beamwidths. Experiments are performed on a small steel needle and blocks of gelatin. To monitor the vibration, a high-frequency probing transducer is used in pulse echo mode with the beam aimed at the vibrating part of the object. This transducer shoots tone bursts at a rate much higher than df. Echoes are quadrature detected and the resulting in-phase and quadrature signals are recorded. These signals are used to calculate the phase of the returned echo which is shown to be proportional to the target displacement. In the experiments, the power beams were driven at 20 W each and $df = 10$ Hz. Amplitude of the resulting vibrations were 5 μm for the gelatin block and 1.2 μm for the needle.

10:00

4aPA8. Nonlinear effects on acoustic field measurement in biomedical frequency range. S. G. Ye, X. F. Gong, and X. Z. Liu (Inst. of Acoust. and Lab. of Modern Acoust., Nanjing Univ., Nanjing 210093, People's Republic of China)

It was shown that ultrasound propagation in media is accompanied with nonlinear phenomena and the nonlinear effects on the measurement of acoustic field cannot be neglected. A needle hydrophone was used to receive the signal in the acoustic field for further analysis and the radiation force method was used to measure the total output power of the transmitter. The case of emitting power was constant, the nonlinear phenomena were accumulated due to the distance between the transmitter, and the hydrophone increased from a few centimeter to less than 20 cm. With increasing distance between the transmitter and the hydrophone, the ultrasound energy was gradually transferred from fundamental frequency f_0 to high harmonics. Using the spectrum analyzer, the dependence of f_0 and nf_0 ($n=2,3,\ldots$) on the distance increased was measured. When the distance between the transmitter and the hydrophone was fixed, the nonlinear phenomena became stronger as the emitting power increased. The acoustic intensity of f_0 was no longer linearly increased with the increased emitting power acoustic intensity, and the acoustic intensity of f_0 would tend toward a saturation value. [Work supported by NSFC.]

10:15

4aPA9. Active acoustic stabilization of capillary bridges significantly beyond the Rayleigh limit: Experimental confirmation. Mark J. Marr-Lyon, David B. Thiessen, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Liquid bridges between two solid surfaces have applications in low gravity such as the solidification of floating zones. Long bridges naturally become unstable to a symmetric mode by bulging near one end while the opposite end thins. For a cylindrical bridge in low gravity of radius R and length L, the slenderness $S=L/2R$ has a natural (Rayleigh) limit of π beyond which the bridge breaks. It has been demonstrated that acoustic radiation pressure may be used in simulated low gravity to produce stable bridges significantly beyond the Rayleigh limit with S as large as 3.6. The bridge (PDMS mixed with a dense liquid) has the same density as the surrounding water bath containing an ultrasonic standing wave. Modulation can be used to excite specific bridge modes [Morse *et al.*, Phys. Fluids **8**, 3–5 (1996)]. The shape of our bridge is optically sensed and the ultrasonic drive is electronically adjusted such that the radiation stress distribution dynamically quenches the most unstable mode. This active control simulates passive stabilization suggested for low gravity [J. Acoust. Soc. Am. **97**, 3377(A) (1995)]. Feedback increases the mode frequency in the naturally stable region. [Work supported by NASA.]

10:30

4aPA10. Acoustic streaming in a resonant channel. Ashok Gopinath (Dept. of Mech. Eng., Naval Postgrad. School, Code ME/Gk, Monterey, CA 93943)

The problem of acoustic streaming in a plane parallel channel supporting a resonant standing acoustic wave is considered. The channel is representative of one of the multitude of channels typically present in the stack of a thermoacoustic engine. The governing equations for the flow field in the channel are developed in a generalized dimensionless form which allows for compressibility effects in the fluid, and also for thermal effects due to wall-fluid coupling to be treated. It is verified that for gaps wide with respect to the thermal penetration depth ($y_0/\delta_\kappa \gg 1$, y_0: channel gap half-width, δ_κ: thermal penetration depth), the classic results of Rayleigh, including the recent modifications to it [Q. Qi, J. Acoust. Soc. Am. **94**, 1090–1098 (1993)], are recovered. However for cases where y_0 and δ_κ are comparable, as in the stack of a thermoacoustic engine, the situation is considerably different and indicates the presence of strong steady streaming flows in the channel which could provide an explanation for the jets experimentally observed at the ends of such a channel as reported at an earlier meeting [Gaitan *et al.*, J. Acoust. Soc. Am. **96**, 3220(A) (1994)]. Steady flows of this type have an important bearing on the design of heat exchangers located at the ends of the stack in a thermoacoustic engine.

10:45

4aPA11. Acoustic sensing of the nonlinear dynamics of density stratified vortices. Alexander B. Ezersky, Poul L. Soustov, and Alexander B. Zobnin (Inst. of Appl. Phys., Russian Acad. Sci., 46 Ulyanov str., Nizhny Novgorod 603600, Russia)

Data on the nonlinear processes in an atmosphere and in the ocean may be obtained by means of remote acoustic sensing. The principal task is to use the data obtained in experiments for reconstruction of temperature, pressure, and velocity fields, and thus reconstruct the flow. Results of laboratory investigations of the potentialities of remote acoustic sensing of flows are presented. The scattering of sound at vortices behind a cylinder was explored. The temperature of the cores of the vortices formed as a boundary layer was detached from the heated cylinder was higher than that of the incident flow. The experiment was performed in a low-turbulence wind tunnel at flow velocities much lower than the sound velocity. It is shown that the characteristics of the scattered signal (indications of scattering and time spectra) may be used to determine vortex parameters such as vorticity, repetition rate, and spatial period, to estimate the temperature of vortex cores and the amount of the heat transferred, as well as to identify the processes of vortex merging and vortex core oscillations. Results of acoustic sensing are in a good agreement with theoretical estimates, direct measurements by a hot-wire anemometer, and data of experiments on vortex visualization.

THURSDAY MORNING, 16 MAY 1996 — REGENCY C, 8:00 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Detection, Discrimination and Masking

Virginia M. Richards, Chair

Department of Psychology, University of Pennsylvania, 3815 Walnut Street, Philadelphia, Pennsylvania 19104

Contributed Papers

8:00

4aPP1. AM detection with one- and two-tone modulators. Stanley Sheft and William A. Yost (Parmly Hearing Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

Psychometric functions were measured for detecting amplitude modulation (AM) of a wideband noise carrier that was either gated with the 400-ms modulator or was on continuously. AM rates were 5, 20, 80, and 2000 Hz. The functions were steepest at the 80-Hz AM rate and shallowest with 2-kHz AM. A linear regression of $\log d'$ on $20 \log m$ showed the slope of the functions ranging from 0.83 to 1.59 in the gated conditions and from 1.34 to 1.79 in the continuous conditions. The increase in function slope between the gated- and the continuous-carrier conditions was less than predicted by Laming's model of sensory processing [D. Laming, *Sensory Analysis* (Academic, New York, 1986)]. AM detection ability was also measured with two-tone modulators centered at either 2 or 6.5 kHz. Modulator tones were separated by 5, 20, or 80 Hz. Performance with the two-tone modulators was often better than predicted by either the envelope

rms or max/min value, especially with the tones centered at 6.5 kHz. This result is consistent with incorporation of a second nonlinearity into a model of auditory envelope detection, introducing intermodulation at the low-modulator beat rates. [Work supported by NIH.]

8:15

4aPP2. Effects of frequency, increment and decrement duration, and pedestal duration on the detection of increments and decrements in sinusoids. Robert W. Peters (Div. of Speech and Hear. Sci. and Dept. of Psych., CB# 7190, Univ. of North Carolina, Chapel Hill, NC 27599-7190) and Brian C. J. Moore (Univ. of Cambridge, Cambridge CB2 3EB, England)

Thresholds for detection of increments and decrements in level of sinusoidal signals were measured as a function of signal duration (10, 20, or 200 ms), pedestal duration before the signal (10, 200 ms, or pedestal on continuously) and frequency (250, 1000, and 4000 Hz). Pedestal duration was varied to study adaptation effects and signal duration was varied to study temporal integration. The sinusoids were presented in a background noise intended to mask spectral splatter. Seven normal-hearing subjects were used. Thresholds improved with increasing frequency and with increased duration for both increments and decrements. Increasing the pedestal duration before the increment from 10 to 200 ms generally improved the threshold for increment durations of 10 and 20 ms but not for an increment duration of 200 ms. When the pedestal was on continuously, distinct adaptation occurred at 4000 Hz; the pedestal appeared to fade away. However, this did not greatly affect increment thresholds, except for a slight improvement for the increment duration of 200 ms. Decrement thresholds generally became worse when the pedestal was on continuously. The results are discussed in terms of the effects of adaptation and temporal integration.

8:30

4aPP3. Detection of change without regard to its valence. Ervin R. Hafter, Anne-Marie Bonnel, and Erick J. Gallun (Dept. of Psych., Univ. of California, Berkeley, CA 94720)

When an increment ($S+$) or a decrement ($S-$) is added to an ongoing pedestal in a *detection* task, performance is far better than when all trials have a signal and the task is to *identify* its valence. This is the opposite of predictions from a standard detection theoretical analysis. It has been suggested that the *detection* task reflects sensitivity to transients associated with stimulus change while *identification* task reflects analysis of the sustained level, i.e., the energy. To be complete, this must argue that detection is of absolute change, regardless of its sign. To test this, detectability was measured with bidirectional signals ($S+$) and ($S-$) was compared that with only a single signal type, either ($S+$) or ($S-$). This was done in paradigms: (1) with the pedestal constant and (2) with gaps in the pedestal on either side of the signal that prevented the use change *per se*. Results show the hypothesis, showing that monodirectional signals were equally detectable as bidirectional signals when there were no gaps but with gaps, they were better.

8:45

4aPP4. Stimulus fluctuations in auditory intensity discrimination. Julius L. Goldstein (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

A basic prediction of Green's [J. Acoust. Soc. Am. **32**, 121–131 (1960)] classical energy-detection theory for Gaussian signals is a variance measure of stimulus fluctuation that decreases monotonically with the signal's duration-bandwidth product (TW). Experiments on intensity discrimination of narrow-band Gaussian noise provide the most direct support of this prediction for $TW<\sim 16$, but indicate a need for another source of fluctuation for larger TW which could reflect internal noise [deBoer, J. Acoust. Soc. Am. **40**, 552–560 (1966)]. An alternative solution is proposed based on detection of the largest waveform peak within the analysis window ($\geq T$). Investigation of this detector for narrow-band Gaussian noise shows that the variance (in dB) of the peak fluctuations is the sum of the variances of the classical energy fluctuation and of a waveform fluctuation due to random phases. For $TW<20$ the energy fluctuation dominates, while the waveform fluctuation dominates for larger TW. Predictions of the classical energy detection model with an internal noise of 1 dB are essentially indistinguishable from the peak detector model for intensity discrimination thresholds of narrow-band Gaussian noise; however, peak detection explains a wider range of experiments [Goldstein, J. Acoust. Soc. Am. **98**, 2907(A) (1995)].

9:00

4aPP5. Mechanisms of FM detection. Brian C. J. Moore and Aleksander Sek (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, England)

In experiment 1, thresholds for detecting FM were measured for carrier frequencies f_c from 0.25 to 6 kHz and for modulation rates f_m from 2 to 20 Hz, using a 2AFC task. To disrupt cues based on changes in the excitation pattern, thresholds were also measured when the carriers in both intervals of a trial were amplitude modulated at the same rate as the FM with a modulation index of 0.333. The phase of the AM relative to the FM was randomized. For $f_c<4$ kHz, the deleterious effect of the AM increased with increasing f_m. For $f_c=6$ kHz, the effect was independent of f_m. In experiment 2, psychometric functions were measured for detecting combined FM and AM, with $f_m=2$ Hz, as a function of the relative phase of the modulators, using modulation depths for AM and FM that would be equally detectable if each were presented alone. Relative modulator phase had no effect for $f_c=0.25$ kHz, small effects for $f_c=1$ kHz, and large effects for $f_c=6$ kHz. The results suggest that a temporal mechanism dominates FM detection for $f_c<4$ kHz, and for $f_m=2$ Hz. A place mechanism dominates for high carrier frequencies, and for lower carrier frequencies when $f_m\geq 10$ Hz.

9:15

4aPP6. Masking additivity in an excitation-pattern model. B. Espinoza-Varas (Communication Sciences & Disorders, Univ. of Oklahoma Health Sciences Ctr., Oklahoma City, OK 73190)

An excitation-pattern model [Moore and Glasberg, Hear. Res. **28**, 209–225 (1987)] was used to predict the masking additivity of two simultaneous sinusoidal maskers. The prediction relies on the assumption that masked threshold is reached when the addition of the signal to the masker produces a critical difference in the masker excitation pattern (EP) at any center frequency (the critical EP difference is termed criterion EPD). For equally effective maskers, masking additivity was assumed be equal to the difference between two signal levels: The level that produces the criterion EPD with the combination of maskers minus the level that produces the criterion EPD with the individual maskers. Excitation-pattern predictions of masking additivity were obtained for signal frequencies of 500, 1000, and 5000 Hz, each masked by two sinusoidal maskers symmetrically placed about the signal frequencies; maskers frequencies within and outside the signal ERB were considered. The masking additivity predicted by the EP model is compared to that predicted by the modified power law model [Humes *et al.*, J. Acoust. Soc. Am. **83**, 188–202 (1988)], and to masking-additivity data [Lutfi, J. Acoust. Soc. Am. **73**, 262–267 (1983)].

9:30

4aPP7. Processing transitions in a band-widening experiment. Bruce G. Berg, Curt Southworth, and Brian K. Branstetter (Dept. of Cognit. Sciences, Univ. of California, Irvine, CA 92717)

Literature suggests that in the absence of single-channel level information, discriminations of narrow-band sounds are temporally based, whereas discriminations of wideband sounds rely on across-channel level comparisons. The transition between processes is investigated using a band-widening paradigm. Thresholds are estimated for increments in the intensity of the central tone of a band of equal intensity, random phase tones evenly separated in frequency (parametrically varied between 10 and 160 Hz). A 20-dB roving-level procedure degrades overall-level information. Uniformly distributed frequency shifts of the complex over a 100-Hz range degrade pitch cues. As the number of tones is increased, thresholds increase up to a certain bandwidth, after which thresholds decrease or remain constant. It is proposed that these breakpoints in threshold functions represent a transition from temporal to spectral discrimination processes. The mean ratio of breakpoints to center frequencies of 500, 1000, 2000, and 4000 Hz is approximately 0.4. Regarding individual differences, a significant positive correlation exists between breakpoints and thresholds in a wideband profile analysis task. [Work supported by ONR.]

9:45

4aPP8. The effects of center frequency and bandwidth on the discriminability of noise. Martin E. Rickert and Donald E. Robinson (Dept. of Psych., Indiana Univ., Bloomington, IN 47405)

When listeners are asked to discriminate between trials on which a sample of noise is presented twice and trials on which two nonidentical samples are presented, discriminations are easier if samples differ near the end rather than near the beginning. The effect of temporal position [S. F. Coble and D. E. Robinson, J. Acoust. Soc. Am. **92**, 2630–2635 (1992)] is robust over a wide range of stimulus parameters including duration, level, and correlation. In previously reported work [M. E. Rickert and D. E. Robinson, J. Acoust. Soc. Am. **98**, 2906(A) (1995)], discriminability was measured with wideband (100–3000 Hz) and narrow-band (455–655 Hz) noise. The size of the effect of temporal position is smaller with the narrow-band noise. However, one could argue that these effects are restricted to the lower region of the frequency spectrum because center frequency was held constant ($\approx$ 545 Hz). In the experiment reported here, both center frequency (545 and 2000 Hz) and bandwidth (200 and 1000 Hz) were investigated. Data were also obtained with wideband (100–3000 Hz) noise. Although the effect of temporal position is consistent across stimulus conditions, the size of the effect is reduced for narrower bandwidths. There is no evidence of an effect of center frequency. Results are discussed in terms of a leaky integrator model [D. E. Robinson and M. E. Rickert, J. Acoust. Soc. Am. **98**, 2906(A) (1995)]. [Work supported by NIH and AFOSR.]

10:00–10:15 Break

10:15

4aPP9. Simulations of temporal resolution and integration data with a computer model using a modulation filter bank. Stefan Münkner (III. Phys. Inst., Göttingen, Germany) and Armin Kohlrausch (Inst. Perc. Res. (IPO), Eindhoven, The Netherlands)

Temporal resolution and temporal integration in the auditory system, with their conflicting time constants, are often regarded as paradox phenomena. In this study a model is presented for the simulation of a number of temporal resolution experiments (e.g., gap detection, forward masking) and classical test tone integration data (cf. contributions by Dau *et al.*). The model consists of two major parts. A preprocessing stage that transforms the acoustic stimulus into an internal representation, simulating the peripheral processing in the auditory system. Processing errors are accounted for by adding internal noise which introduces detection limits. The second stage is an optimum detector to simulate a 'central' decision device. The preprocessing stage consists mainly of three subunits: a Gamma tone filterbank to simulate peripheral filtering; a set of adaptive feedback loops that derives an adapted envelope of every peripheral filter output; a modulation filterbank (per peripheral channel) that analyzes stimulus periodicities in these adapted representations. The model is able to simulate gap detection and test tone integration experiments quantitatively without changing its parameters. It can be shown that within this model the modulation filterbank is vital to achieve this performance. [Supported by the DFG (Az. Mu/1201).]

10:30

4aPP10. Toward a simple resolution of the temporal resolution/integration paradox. C. Formby (Div. of Otolaryngol.-Head & Neck Surgery, Univ. of Maryland School of Medicine, 16 S. Eutaw, Ste. 500, Baltimore, MD 21201), T. G. Forrest (Cornell Univ., Ithaca, NY), M. G. Heinz (Harvard–MIT, Cambridge, MA 02139), S. E. Hargus, and J. W. Zeiders (Univ. of Maryland School of Medicine, Baltimore, MD 21201)

Green [*Time Resolution in Auditory Systems*, 1985] distinguished between two broad classes of paradigms that presumably represent the two ends of the auditory temporal processing continuum. At one extreme are measures of temporal integration (TI) that estimate the maximum time over which acoustic information can be stored. At the other extreme are measures of temporal acuity/resolution (TA) that estimate the briefest detectable change in acoustic information. The goal of both paradigms is to estimate a time constant (τ) for the auditory system. Traditionally, τ_I reported for TI is more than an order of magnitude longer than τ_A estimated for TA. The aim of this study was to resolve this apparent paradox. Detection thresholds were obtained for a range of brief bandlimited (W_N=62–6000 Hz) increments within a broadband noise to estimate τ in TI and TA tasks from the same listeners. For each W_N condition, detection thresholds were measured as a function of (1) increment duration (P_N=10–480 ms) by tracking increment level adaptively to estimate τ_I and (2) increment level by tracking increment duration adaptively to estimate τ_A. The resulting τ_I and τ_A estimates are comparable and vary inversely with W_N from about 70 to 7 ms. [Research supported by NIH.]

10:45

4aPP11. Fast-acting compression in human hearing. Robert P. Carlyon and A. Jaysurya Datta (MRC Appl. Psych. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, England)

Physiologically motivated auditory models calculate excitation from a *compressed* and rectified version of auditory filter responses. Unlike the "power spectrum" model, they predict that, when the responses of a filter to two sounds have equal power but different peak factors, the sound producing the less peaked response will produce more excitation. This was confirmed by experiments using two sounds, each consisting of harmonics 2–20 of a 100-Hz fundamental, presented at 68 dB SPL/harmonic, and with the harmonics summed in either positive or negative Schroeder phase. Kohlrausch and Sander [J. Acoust. Soc. Am. **97**, 1817–1829 (1995)] have shown that the response of an auditory filter centered on 1100 Hz to the positive phase complex is highly modulated, whereas that to the negative phase complex is not. It is reported that the threshold for a 10-ms 1100-Hz signal presented 5 ms after the positive phase stimulus is approximately 10 dB less than after the negative phase stimulus. Also, listeners judged harmonics 9–13 of the positive phase sound to be quieter than the corresponding harmonics of the negative phase sound. It is concluded that the results reflect fast-acting compression in the human auditory system.

11:00

4aPP12. Peripheral origins of the upward spread of masking. Andrew J. Oxenham (Inst. for Percept. Res. (IPO), P.O. Box 513, 5600 MB Eindhoven, The Netherlands) and Christopher J. Plack (Lab. of Experimental Psych., Univ. of Sussex, Brighton BN1 9QG, England)

Recent physiological studies have shown that the basilar membrane (BM) response to tones at characteristic frequency (CF) is highly compressive at medium levels, with a ratio of about 5:1, while the response to tones well below CF is linear. A psychophysical measurement of these characteristics was attempted using forward masking, to avoid peripheral interaction of stimuli, with a 4-ms, 6-kHz signal presented 2 ms after the offset of a 100-ms, 3-kHz masker, in order to measure thresholds at high signal levels. The growth of masking was very compressive at medium signal levels; increasing the signal level from 50 to 70 dB SPL resulted in a mean increase in masker level of only 3.6 dB. Growth at lower and higher signal levels was more linear. Thresholds were also measured for a number of other masker frequencies. As expected, the difference between masker and signal growth rate decreased as the masker frequency approached signal frequency. The results are quantitatively consistent with the hypothesis that BM nonlinearity governs the upward spread of masking. Furthermore, the technique may provide a simple way of estimating BM nonlinearity in humans. [Supported by the Wellcome Trust and the Royal Society.]

11:15

4aPP13. Basilar membrane nonlinearity and the growth of forward masking. Christopher J. Plack (Lab. of Experimental Psych., Univ. of Sussex, Brighton BN1 9QG, England) and Andrew J. Oxenham (Inst. for Percept. Res. (IPO), P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

Psychophysical forward masking generally grows nonlinearly, with a given increase in signal level requiring a larger increase in masker level for the signal to remain at threshold. In the present experiments, a 10-ms sinusoidal signal was presented 2 ms after a 20-ms sinusoidal masker at the same frequency. Frequencies of 2 and 6 kHz were tested. It was shown that, while the masker level needed grew faster than the signal level at low signal levels, at medium signal levels the reverse was the case. These results can be understood in terms of a basilar membrane response that is compressive at medium levels and more linear at low and high levels. When the masker is in the compressive region and the signal is in the lower, more linear, region, then large increases in masker level are needed for a small increase in signal level. When the signal is in the compressive region and the masker is in the higher, more linear, region, then large increases in signal level are needed for a small increase in masker level. [Work supported by the Royal Society and the Wellcome Trust.]

11:30

4aPP14. Psychophysical evidence of excitatory connections across auditory frequency channels. Beverly A. Wright (Keck Ctr., P.O. Box 0732, Univ. of California, San Francisco, CA 94143-0732)

A simultaneously masked auditory signal becomes easier to detect as its onset is delayed from masker onset. This "enhancement effect" depends upon masking components presented outside of the signal's frequency channel, indicating the participation of an across-channel process. The nature of this process has not been clearly established due to the difficulty of separating the influence of excitation and inhibition in simultaneous masking. Here, the forward-masked threshold of a 20-ms, 1000-Hz tone was determined in the presence of one or two masking tones [M1 (1000 Hz, 50 dB) and M2 (variable from 500 to 3000 Hz, 70 dB)]. When the masker duration was 500 ms, adding M2 to M1 reduced signal threshold when M2 was presented at 1150 Hz (suppression), but had no influence on performance when M2 was above about 1300 Hz. In contrast, when the masker duration was 40 ms, adding M2 to M1 did not lower threshold at 1150 Hz, and always increased threshold when M2 was presented above 1300 Hz. This latter threshold increase was reduced when a third masking tone presented at 1150 Hz was added to M1 and M2. These results suggest that the across-channel process involved in the enhancement effect is excitatory, adapts, and is moderated by suppression. [Work supported by NIDCD.]

11:45

4aPP15. New masking experiments using low-noise noise maskers. Armin Kohlrausch (Inst. Percept. Res. (IPO), P.O. Box 513, NL-5600 MB Eindhoven, The Netherlands)

Low-noise noise generally refers to narrow-band noise with a flat temporal envelope. Hartmann and Pumplin [J. Acoust. Soc. Am. **83**, 2277–2289 (1988)] used one 100-Hz-wide low-noise noise sample to mask a sinusoidal signal, and found that thresholds were 5 dB lower than for a comparable Gaussian noise. These experiments were performed as frozen-noise measurements and the threshold values were the averages of the results for six different signal starting phases. The thresholds for the signal in Gaussian noise (about −11-dB signal-to-overall-noise ratio) were considerably lower than thresholds typically found in a random-noise presentation (−3 dB). The experiments reported here were initiated to test whether the difference in masking behavior between low-noise noise and Gaussian noise is also observed in a random-noise experiment and how it depends on masker bandwidth and signal duration. In a 100-Hz-wide masker centered at 1000 Hz, the same difference of about 5 dB was found. This difference increases with decreasing masker bandwidth, reaching a maximum of 9.5 dB for a 10-Hz-wide masker. The difference also increases somewhat with decreasing signal duration, from 5 dB for a 500-ms signal to 7.6 dB for a 20-ms signal.

THURSDAY MORNING, 16 MAY 1996 CELEBRATION A, 8:15 TO 11:45 A.M.

Session 4aSA

Structural Acoustics and Vibration: Scattering

Guillermo C. Gaunaurd, Chair

Carderock Division, Naval Surface Warfare Center, White Oak, Code 684, Silver Spring, Maryland 20903-5640

Contributed Papers

8:15

4aSA1. Angular dependence of Bloch wavepackets. Charles F. Gaumond and David Drumheller (Naval Res. Lab., Code 7140, Washington, DC 20375-5350)

The contribution to an echo by a particular scattering mechanism, which occupies a specific region in the time-frequency or time-scale domains, is defined as a wavepacket [N. Yen, J. Acoust. Soc. Am. **81**, 1841–1850 (1987)]. Previous studies have isolated wavepackets generated by specular reflection, Bragg reflection and Bloch waves in backscattered echoes from a ribbed, finite cylinder at a few discrete target aspect angles. For this target, the echoes mainly contain components from specular reflection and Bloch waves in the frequency region near $ka \simeq 5$. These two kinds of wavepackets are experimentally extracted in the time-scale domain over the angular range from 10° to 70°. The time extents and spectra of these wavepackets are shown to vary with target aspect angle. This angular dependence is shown to be related to physical mechanisms. [Work supported by NRL 6.2 funds.]

8:30

4aSA2. (Bragg mode)-(leaky mode) coupling due to a finite periodic array of scatterers on a submerged elastic layer. Raymond J. Nagem, Leopold B. Felsen, and Brian J. Collins (Dept. of Aerospace and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Coupling of energy into and out of submerged layered elastic topographies via scattering centers on the top surface is of interest for a variety of applications in bottom-interacting acoustics. To understand and parametrize the basic wave phenomena which are operative here, this study was begun with a simple model wherein a finite periodic array of scatterers situated on a single elastic layer is insonified by an incident plane wave. The problem is analyzed rigorously by spectral wave-number decomposition, and the scattered fields are obtained by asymptotic evaluation of the spectral intergrals. The results can be interpreted in terms of array-modified leaky waves and slab-modified Bragg modes. with Bragg-leaky mode coupling occurring due to end effects as well as bulk effects of the array. The coupling effects are optimized when one of the slab-modified modes is phased matched to one of the Bragg modes. The asymptotic algorithms are calibrated against numerically evaluated spectral integral reference solutions. Windowed space-wave-number processing is then applied to extract the above phenomenology from the reference data.

8:45

4aSA3. First circumferential wave in the low-frequency scattering by a viscoelastic cylinder. Farid Chati, Fernand Leon, and Gérard Maze (Lab. d'Acoust. Ultrason. et d'Électron., LAUE URA CNRS 1373, Univ. du Havre, Pl. Robert Schuman, 76610 Le Havre, France)

Acoustic scattering from an absorptive cylinder (Lucite) of infinite length excited by a normal plane acoustic wave is investigated in low frequencies. This one is assumed to be homogeneous and isotropic in its viscoelastic properties. The experimental results are obtained with a method of isolation and identification of resonances (MIIR). The resonance spectra are particularly studied. In the given frequency range, the presence of six families of circumferential waves labeled l_w ($1 \leq l_w \leq 6$) was noted. The resonances of these families, except $l_u = 1$, are wide compared to those of aluminum or steel cylinders. Nevertheless, the resonances of the first family are easily detected in the Lucite cylinder, contrary to aluminum or steel cylinders. The scattering of a plane-wave pulse can be represented by the Fourier integral in order to determine a theoretical resonance spectrum without choosing the background. A good agreement is obtained for the studied frequency range between experimental and theoretical results. In the case of aluminum or steel cylinders, the computed calculus in vacuum are in good agreement with the experimental results. In the case of the Lucite cylinder, that is not verified, particularly with the first family. Moreover, the effect of the external fluid density on the resonances is analyzed.

9:00

4aSA4. Acoustic scattering by a fluid-loaded cylindrical shell with an internal plate. Aleksander Klauson, Jaan Metsaveer (Dept. of Mechanics, Tallinn Tech. Univ., Ehitajate tee 5, EE-0026 Tallinn, Estonia), André Baillard, Dominique Decultot, and Gérard Maze (Univ. du Havre, 76610 Le Havre, France)

An experimental and theoretical study of acoustic scattering from a fluid-loaded circular cylindrical shell with an axially attached internal plate is presented. The normally insonified stainless-steel shell has an outer to inner radius ratio $b/a=0.98$. A stainless-steel plate of the same thickness is diametrically soldered, inside the shell, at both edges along the axis. The object is closed by two flat circular endcaps and is vertically immersed in a water tank (6 m$\times$4 m$\times$3 m). The aim of the study is to separate the effects of the shell and those of the plate and to interpret different sound generation mechanisms of the structure. The numerical simulation is done by the modal expansion method using the unstiffened cylindrical shell eigenmodes. The shell is taken as infinitely long. The internal plate is considered as a line constraint with the frequency-dependent impedance. The comparison between theoretical and experimental results is achieved by an analysis of backscattered spectra and of arrival times of backscattered echoes. The scattering from the structural joints, the generation and transmission of flexural waves through the internal plate, and the interaction between different vibration forms are observed and analyzed. [Work supported by NATO.]

9:15

4aSA5. Backscattering due to ray reverberations between nonconcentric surfaces of a thick shell: Ray stability, transitions in wavefront shapes, and observed scattering enhancements due to focusing. Scot F. Morse and Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

The paraxial ray stability analysis previously developed for laser resonators is applied to high-frequency elastic wave reverberations across the thickness of a shell having nonconcentric surfaces. A previous summation of backscattered amplitudes from rays reverberant within concentric surfaces [S. G. Kargl and P. L. Marston, J. Acoust. Soc. Am. **88**, 1114–1122 (1990)] is found to lie on the boundary between stable and unstable resonators. Shifting the center of the inner surface of the shell away from the direction of the source produces a stable resonator. Rays reverberant between such nonconcentric surfaces produce a far-field focusing that is periodic in the number of internal bounces. Exact ray tracing, used to identify scattered wavefront shapes near focal conditions, reveals the presence of glory rays (off-axis rays which are exactly backscattered). Shifting the inner surface of the shell toward the source produces an unstable resonator and no focused rays. The existence of backscattering enhancements by focused reverberant rays for a stable resonator was confirmed with experimental observations with elastic scatterers in water having nonconcentric spherical surfaces. [Sponsored by the Office of Naval Research.]

9:30

4aSA6. Scattering by elastic elliptical cylinders of different aspect ratio. Jacques Lanfranchi and Gérard Maze (Lab. d'Acoust. Ultrason. et d'Electron., LAUE URA CNRS 1373, Univ. du Havre, Pl. Robert Schuman, 76610 Le Havre, France)

The scattering of elliptical cylinders of infinite length insonified by an incident beam perpendicular to their axis is investigated. The method of isolation and identification of resonances (MIIR) allows one to plot resonance spectra. These spectra are obtained from three targets characterized by their major radius/minor radius ratio equal to $\frac{4}{3}$, $\frac{4}{2}$, $\frac{4}{1}$. The resonance spectra obtained from a circular cylinder excited in the same conditions show peaks related to circumferential waves which, for particular frequencies, constitute standing waves. In this case, the phase velocity depends only on the frequency. Contrary to the circular cylinder, resonance spectra obtained by an experimental monostatic method depend on the azimuthal position of the transducer. Some resonances vanish at certain positions. In this case, the phase velocity and the coupling coefficient are a function of the curvature radius and the frequency. The wavelength is not identical on the circumference of the elliptical cylinder. To explain the experimental results, a phase-matching model is developed to determine the resonance frequencies [H. Überall *et al.*, J. Acoust. Soc. Am. **81**, 312–315 (1987)]. This method allows one to represent the vibration state on the circumference. An integral calculus applied on the latter result gives the far-field pressure.

9:45

4aSA7. High-frequency backscattering by thick finite cylindrical shells in water at oblique incidence: Experiments and calculations. Scot F. Morse, Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814), and Gregory Kaduchak (Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Various authors have demonstrated the importance of plotting backscattering amplitude as a function of both aspect angle and frequency for thin cylindrical shells in water. Some interesting features of analogous plots for a thick finite cylindrical shell with the frequency range extending above the coincidence region have been investigated. Impulse response measurements taken with an improved PVDF sheet source [Kaduchak *et al.*, J. Acoust. Soc. Am. **97**, 2699–2708 (1995)] reveal a backscattering enhancement in the coincidence frequency region for aspect angles in the vicinity of end-on incidence. Though the cylinder was capped, the features of interest appear to be identified with waves on the coupled shell–fluid system. Similar features are present in backscattering computations for a simply supported thick cylindrical shell and they may be useful in high-frequency inverse scattering. [Sponsored in part by the Office of Naval Research and by the UT:ARL Independent Research and Development Program.]

10:00

4aSA8. Strong bending modes for scattering from elastic spheroidal solids and shells. Michael F. Werby (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529) and Nataly A. Sidorovskaia (Univ. of New Orleans, New Orleans, LA 70148)

Bending (flexing) modes are excited when signals scatter at oblique incidence from spheroidal shells [M. Werby *et al.*, J. Acoust. Soc. Am. **85**, 2365 (1989)]. This was inferred by comparing the exact T-matrix resonance calculations with those predicted from Timoshenko beam theory. A more sophisticated approach was also developed using the general phase-integral method that allowed for nonuniform shapes. The possibility of the existence of such modes not only for fairly thick shells but also for thin shells is considered. Acoustical signals scattered by elastic shells of various materials, aspect ratios and shell thickness are investigated. The presence of bending modes even for very thin shells is demonstrated. It is showed that for thick shells the resonances manifest themselves as maximum amplitude returns while for thin shells they manifest themselves as minimum amplitude returns. The effect is associated with the transitional nature of a rigid-like background to a soft-like background for the two extremes so that the return signals vary in their coherence from adding constructively to adding destructively over the thickness variation. The sensitivity of resonance locations as a function of the elastic parameters is also presented. [Work supported by Naval Research Laboratory and Office of Naval Research.]

10:15–10:30 Break

10:30

4aSA9. Theory of the scattering of an acoustical signal from an elastic spheroidal shell near reflecting or absorbing interfaces. Nataly A. Sidorovskaia (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148) and Michael F. Werby (Naval Res. Lab., Stennis Space Center, MS 39529)

A new formulation that allows one to describe backscattered echoes from elastic shells near pressure release, rigid, and absorbing planer interfaces is presented. The proposed formulation is always consistent—in contrast to earlier formulations—even at low frequencies and large distances from the interface. It allows for the rapid reproduction of backscattered echoes over frequency ranges of practical interest. This novel approach is developed and is shown to be sufficiently general to allow for nonreflecting smooth surfaces. The method has been implemented and numerical results for both pulse and continuous wave signals scattering from various shells are carried out. The studies for various object materials, distances from the interface, and for the effect that the three type of interfaces have on the detected signals are presented. [Work supported by Naval Research Laboratory and Office of Naval Research.]

10:45

4aSA10. Diffraction from simple shapes by a hybrid asymptotic-numerical method. Joshua M. Montgomery and Paul E. Barbone (Dept. of Aerospace & Mech. Eng., Boston Univ., Boston, MA 02215)

The application of a hybrid asymptotic/finite-element method to the problem of scattering from prismatically shaped objects is considered. The hybrid method is based on patching a short wavelength asymptotic expansion of the scattered field to a finite-element interpolation of the near field. In patching, the diffracted field shape functions with unknown amplitude are forced to agree smoothly with the solution in the near field along a curve at a prescribed distance from the diffraction points. An asymptotic DtN (Dirichlet-to-Neumann) map on this artificial boundary represents the effect of the outer domain on the solution within this new boundary. This allows us to replace the original boundary value problem with an asymptotically equivalent boundary value problem, the domain of which is small and efficiently discretized. The method is applied to diffraction by a blunted wedge, which in this context represents a degenerate prism. The hybrid scattering solution shall be compared to a complete analytic field representation found using matched asymptotic expansions. [Work supported by ONR.]

11:00

4aSA11. Acoustical scattering by a density contrast wedge. Anthony M. J. Davis (Dept. of Math., Univ. of Alabama, Tuscaloosa, AL 35487-0350) and Robert W. Scharstein (Univ. of Alabama, Tuscaloosa, AL 35487-0286)

Consider the three-dimensional scattering of a sound pulse generated by an impulsive point source and incident upon a penetrable wedge, identified by a density contrast. The wave speed is common to both regions and the radiation condition of only outgoing waves at infinity is applied in all directions. At the boundary of the wedge there is a pair of transmission conditions which ensure continuity of the acoustic pressure and normal velocity. By using Fourier transforms in time and parallel to the wedge generators and a Kontorovich–Lebedev transform in the radial direction, as described by Jones [*Acoustic and Electromagnetic Waves*, Oxford (1986)], both the exterior and interior fields can be obtained as a sum of impulsive terms, some of which are due to edge diffraction. If the wedge angle is a rational fraction of π, then the residue series can be summed and, after careful consideration of when and where impulsive disturbances can occur, the fields can be written in remarkably simple closed forms. This solution provides the zero-order field for a relatively small difference in wave speeds.

11:15

4aSA12. Backscattering of obliquely incident plane waves by a composite cylindrical shell constructed of isotropic and transversely isotropic layers. Gregory Kaduchak and Charles M. Loeffler (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Acoustic scattering from a transversely isotropic cylindrical shell excited by an obliquely incident plane wave is examined. The shell is comprised of N layers which may be described by isotropic or transversely isotropic material parameters. The present research solves the boundary value problem for a transversely isotropic, infinite cylindrical layer within the framework of exact 3-D elasticity theory. The layers which comprise the shell are connected via a "propagator matrix" which relates the interior and exterior boundary conditions. The backscattering form function is then constructed for several commonly used composite materials which display transverse isotropy. The results for anisotropic shells are compared to results found for isotropic shells with similar parameters. Here, attention will be given to the similarities (and dissimilarities) of the scattering mechanisms which are the chief contributors to the backscattering form function as the degree of transverse anisotropy is increased. Analysis of the scattered waveform through Fourier synthesis into the time domain will also be discussed. [Work supported by ONR.]

11:30

4aSA13. Inverse scattering using backpropagated eigenfunctions. T. Douglas Mast (Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627), Adrian I. Nachman (Univ. of Rochester, Rochester, NY 14627), and Robert C. Waag (Univ. of Rochester, Rochester, NY 14627)

A new approach to inverse scattering using the eigenfunctions of scattering operators is presented. This approach provides a unified framework that encompasses eigenfunction methods of focussing, diffraction tomography, and inverse scattering in arbitrary media. The eigenfunctions of scattering operators are used to specify source distributions that focus incident energy in the vicinity of inhomogeneities. The present inverse scattering approach represents unknown scattering media using products of numerically backpropagated fields of eigenfunctions. Recent progress in the mathematical theory of inverse scattering has suggested that these products are an appropriate basis for the reconstruction of unknown inhomogeneities. Because of the focussing property of scattering operators, these products also form an efficient basis for computational implementations of inverse scattering. The currently suggested approach has the further advantage of applicability to any medium for which a background Green's function can be determined. Computational results illustrate focussing of eigenfunctions on discrete and distributed scattering media, quantitative imaging analogous to diffraction tomography, and high-resolution inverse scattering in media with strongly scattering inhomogeneous backgrounds.

THURSDAY MORNING, 16 MAY 1996 MT. RAINIER AND MT. MCKINLEY, 8:30 TO 11:30 A.M.

Session 4aSC

Speech Communication: Potpourri (Poster Session)

James M. Hillenbrand, Chair
Department Speech Pathology and Audiology, Western Michigan University, Kalamazoo, Michigan 49008

Contributed Papers

All posters will be on display from 8:30 to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 to 11:30 a.m.

4aSC1. Temporal coordination of articulator gestures. Sidney A. J. Wood (Dept. of Linguist., Univ. of Lund, Helgonabacken 12, 22362 Lund, Sweden)

Movement data are analyzed to elucidate principles of articulator coordination. Procedures for tracking articulator gestures from x-ray motion films, and results obtained from Swedish and Bulgarian work are reported in Wood [J. Phon. 19, 281–292 (1991), Proc. 3rd Congress I.C.P.L.A., 191-200, Helsinki (1994), Proc. 13th I.C.Ph.Sc., Vol. 1, 392–395, Stockholm (1995), J. Phon. (in press)]. Coarticulation and gesture conflict have been studied. The domain of coarticulation has been seen variously as a between-target transition, or comprising of one or two phonemes on either side. These are usually presented as rival solutions, but the present data show speakers using all three schemes. The pertinent question is how does a speaker select one of them. For gesture conflict, there are two approaches. The one feeds all gestures to the musculature where conflicts are resolved by summing rival forces. The second modifies motor input by queuing antagonistic gestures in order to avoid conflicts. All potential gesture conflicts in the present data were resolved by queuing gestures. The data to be reported here include Eskimo to provide a further test on the universality of these principles for articulator coordination. [Work supported by Swedish Research Council for Humanities and Social Sciences.]

4aSC2. Articulatory model for female and child speakers. T. V. Ananthapadmanabha (Voice and Speech Systems, 53, Temple St., Malleswaram, Bangalore 560003, India)

Articulatory models reported in the literature are generally meant for a prototype male speaker. One such popular model is that proposed by Mermelstein [J. Acoust. Soc. Am. 53, 1070–1083 (1973)]. In this paper the scope of Mermelstein's model is extended to adopt it for any male speaker as well as for female and child speakers. The default vocal tract shape of the model for a given vowel is determined. The axis of the vocal tract is divided into a number of sections. Vocal tract length as well as section lengths are determined. To adopt the model for a male speaker with a vocal tract length different from that of the model, the section lengths are uniformly scaled. To adopt the model for a female or child speaker, the section lengths are scaled disproportionately over the pharyngeal and oral cavity to attain the speaker's vocal tract length. The disproportionate scale factors are based on the anatomical data. Accuracy and application of the extended model are discussed.

4aSC3. Acoustic correlates of fricatives and affricates. Wendy A. Castleman and Randy L. Diehl (Univ. of Texas at Austin, Austin, TX 78713-8029)

Six adult speakers of American English produced voiceless fricatives and affricates in word initial position in a carrier phrase, varying vowel context, and stress patterns. From each of the productions, the following acoustic measurements were made: silence duration, frication duration, amplitude rise time, amplitude rise slope, frication onset centroid frequency, and following vowel duration. A discriminant function analysis revealed that silence duration was the best at discriminating between the fricatives and affricates, followed by frication duration, onset frequency, and rise slope. No additional discrimination was found due to either rise time or vowel duration. The traditional acoustic measure of rise time, which includes the duration and the slope of the rise, may be due entirely to the contribution of the slope rather than the duration itself. Although rise time has long been considered an important perceptual cue in the distinction between fricatives and affricates, recent research has brought this into question [K. R. Kluender and M. A. Walsh, Percept. Psychophys. 51, 328–333 (1992)]. The results of this study suggest that duration measurements (preceding silence interval and frication duration) are more salient acoustic correlates than amplitude rise-time measures. [Work supported by NIDCD.]

4aSC4. Abstract withdrawn.

4aSC5. Study of the acoustic-to-articulatory transformation for vowels and fricatives. Edward L. Riegelsberger and Ashok K. Krishnamurthy (Dept. of Elec. Eng., Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210)

Using synthetic speech from an articulatory speech synthesizer, statistics are generated of the error between actual articulatory configurations and those estimated by an acoustic-to-articulatory mapping routine. Based solely on acoustics, neglecting aerodynamic and perceptual issues, histograms of total estimation error suggest that the inverse problem is no more ambiguous for fricatives than for vowels. By examining the error covariance, dominant articulatory dimensions are identified in the fricative model that have the greatest effect on the acoustic transfer function and, as a result, are better estimated by the acoustic-to-articulatory mapping routine. Weak articulatory dimensions are also found that the acoustic-to-articulatory mapping routine can barely estimate better than simply guessing. Suggestions are made for ways in which these error statistics, and specifically the knowledge of the unequal importance of different articulatory dimensions, can be used to motivate improved techniques for the acoustic-to-articulatory mapping of speech. Demonstrations of some of these ideas will be given on real and synthetic speech. [Work supported by the AFOSR.]

4aSC6. Fundamental frequency in minimal pair sentences. J. D. Trout (Philosophy Dept. and the Parmly Hearing Inst., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The declination of fundamental frequency was investigated in minimal pair sentences which sound alike but which have word boundaries in different positions (e.g., The monkey races it/The monk erases it). Early results suggest that fundamental frequency declines over these minimal pair sentences in accordance with the results reported on global declination effects [t'Hart *et al.*, *A Perceptual Study of Intonation* (Cambridge U. P., New York, 1990)]. Baseline fundamental frequency over sentences will be compared to that for word boundaries in two-word sequences (e.g., monkey races/monk erases) excised from those minimal pair sentences. Further analysis will determine whether local properties of fundamental frequency movement and acceleration differ significantly in word boundary versus intraword position, and so whether such properties may act as a local cue to word boundary. [Work supported by NIDCD Program Project Grant.]

4aSC7. Effects of "clear speech" on duration-dependent vowel contrasts. Jean L. DeMerit (Linguist. Dept., Univ. of Wisconsin—Madison, 1220 Linden Dr., Madison, WI 53706)

Acoustic and perceptual effects of "clear speech" were investigated by focusing on two duration-dependent vowel contrasts in American English, /ɛ, æ/ and /ʌ, ɐ/. Minimal pairs of words were recorded in carrier sentences by four native female speakers under two speech style conditions, conversational and clear. Measurements of vowel duration and formant frequency were made. The acoustic analyses revealed patterns of durational behavior that differed by speaker but were generally uniform across both vowel contrasts. $F1$ and $F2$ frequencies from the grouped data were displaced toward more peripheral regions of the vowel space in clear speech. However, no consistent patterning was evident across individual speakers. A perceptual test using gated speech stimuli from two of the speakers was used to assess the relative amount of acoustic signal required to identify each vowel per speech style. Also, vowel identification rates were examined for initial gated segments (which included the first 25 ms of the vowel) as well as for full word presentation. Statistical analyses showed significant effects of speech style on vowel identification. Interactions among several factors, however, indicate that the perceptual improvements from speaking clearly depend on the speaker, vowel category and subsequent consonantal context.

4aSC8. Prosody as a potential cue for age perception. James V. Ralston, George Fidas (Dept. of Psych., Ithaca College, Ithaca, NY 14850), Christin E. Agli, and Colleen Coppola (Ithaca College, NY 14850)

Previous research has shown that relatively localized acoustic information, such as voice pitch, $f0$ perturbation, and hoarseness are cues to age perception. The present studies examined the potential role of sentence-level prosodic information in the perception of age. Sixteen different six-word sentences were recorded from eight different males in each of three age groups (middle school students, college students, and senior citizens). Several acoustic measurements were made of each sentence, including mean frequency, slope of intonation contour across entire sentence, slope of terminal fall, and amount of inflection (i.e., absolute frequency change summed across the sentence). A second set of sentences was constructed by randomly reordering the words in each sentence. Original sentences and their scrambled counterparts were presented to 19 listeners who rated the age in years of the speaker of each sentence. Analyses of variance revealed that age judgments were less extreme for scrambled sentences as compared to the regular sentences, particularly for the college-aged and elder voices. These results suggest that the scrambling altered sentence-level vocal information. Preliminary regression analyses also indicate that perceived age is related to some relatively global acoustic measures, such as amount of inflection.

4aSC9. Rounding and palatalization convergence in Twi. Kenneth J. de Jong and Samuel Gyasi Obeng (Dept. of Linguist., 322 Memorial Hall, Indiana Univ., Bloomington, IN 47403)

This paper reports an acoustic and articulatory study of labio-palatalization in Twi Akan (West African). Labio-palatalization is rare cross-linguistically, as would be predicted from theories of vowel dispersion, since palatalization and labialization have opposite effects on second formant frequency. Labio-palatalization in Twi, however, is the ongoing result of a convergence of contrastive consonant rounding and vowel-induced palatalization which have been functionally combined in certain segments. Preliminary palatographic data indicate that rounded coronal consonants have more posterior coronal constrictions. Present acoustic analyses show rounded consonants to be marked by a lowering of $F3$. Combining coronal retraction and rounding, thus has two effects. (1) It produces a stable region of low $F3$, as indicated by nomograms in Wood (1986). (2) It increases front cavity length, resulting in a lower and more compact spectrum of consonantal frication noise. Thus 1) examining the role of uncommon contrasts in a phonemic system can shed light on posited mechanisms for language change, and 2) rounding may serve different roles in different language systems, to lower both front and back-cavity resonances, or a more specific role of lowering front-cavity resonances which shape consonantal noise.

4aSC10. The acquisition of Taiwanese (Amoy) initial stops. Ho-hsien Pan (Dept. of Foreign Languages and Literature, Chiao Tung Univ., Hsinchu, 30050, Taiwan)

Taiwanese (Amoy) is one of the few Chinese dialects with a three-way contrast. There are three Taiwanese syllable initial voiceless unaspirated stops, /p,t,k/, three voiceless aspirated stops, /p^h,t^h,k^h/, and two voiced stops, /b,g/. This paper reports on two studies of the acquisition of the voicing contrast in Taiwanese. The longitudinal study followed two girls from about 28 months to 33 months and to 40 months, respectively. The cross-sectional study compared VOTs in 54 children ranging from 30 months to 6 years. In the longitudinal study, the children had already begun to acquire the contrast between the two voiceless stop types in the begin-

ning of recording, although there were still many tokens of voiceless aspirated stops produced with short lag VOTs. Later, the VOTs of the voiceless aspirated stops were hyperaspirated with VOTs exceeding the adult norm, and there were fewer tokens with short lag VOTs. Both children began to acquire the voiced stops around 33 months. The cross-sectional study confirm Kewley-Port and Preston's (1974) claim and shows further that 6-year-olds' production of voiced stops is not completely adult-like.

4aSC11. The consonantal properties of the unrounded high-mid back vowel /ɤ/ in Mandarin Chinese. Jie Zhang (Dept. of Linguist. Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095)

The vowel /ɤ/ in Mandarin has traditionally been documented as a simple vowel although it has been noticed to involve more complicated articulator movements in its production than a monophthong. Three male speakers of Beijing Mandarin were recorded for an acoustic study of this vowel. The frequency and energy values of F1 through F4 of the beginning part (around 20 ms), the first quarter, and the rest of the vowel were measured from the spectrogram and LPC using CSL. Results of two-tail paired t tests show that the first quarter of the vowel has significantly weaker energy in F2 through F4 than the rest of the vowel at $p=0.05$ level, indicating approximantlike properties in this part. Also, the initial 20 ms of the vowel has higher F1 and lower F2 than the following part, indicating a possible pharyngeal constriction. So phonetically, /ɤ/ consists of a very short pharyngeal approximant [ʕ], a velar approximant [ɰ], and a real vowel [ɤ] in sequence. This phonetic characteristic accounts for the fact that glottal stop insertion before a syllable-initial /ɤ/ is much less likely than that before a real onsetless monophthong /a/, but patterns with that before a syllable-initial /w$_c$/.

4aSC12. Perception of English /r,l,n,d/ by native speakers of Chinese. Anna Marie Schmidt (School of Speech Pathol. and Audiol., P.O. Box 5190, Kent State Univ., Kent, OH 44242)

Speakers of some southern dialects of Chinese do not make a distinction between /n/ and /l/ in Chinese. Previous pilot research indicated that they also have difficulty with this distinction both perceptually and productively in English. As a basis for an experiment in training this distinction, a closer exploration of these speakers' perception of English was performed. This study examined these speakers' perception of English /r,l,n,d/. Five native Chinese speakers heard English words from minimally different sets produced by three native English speakers and labeled the first consonant of each word. Native English speakers labeled the same sets with high accuracy. Results and implications for training will be presented.

4aSC13. Effects of speech style on the perceptual assimilation of American English vowels by Japanese speakers. Reiko A. Yamada (ATR, 2-2, Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-02 Japan), Winifred Strange, Brett H. Fitzgerald (Univ. of South Florida, Tampa, FL 33620), and Rieko Kubo (Nara, 631 Japan)

Of specific interest in this study was the extent to which speech style (citation form versus sentence form) affected perceptual assimilation of AE vowels by Japanese listeners. Four adult male speakers produced 11 AE vowels in /hVba/ bisyllables in both citation form (lists) and in the sentence "I hear the /hVb/ on the tape." Response categories, designated in Katakana symbols, included five short vowels, five long vowels, and eight vowel combinations (e.g., ou, ei, ja:). Twenty-four Japanese listeners selected the response category which was most similar to the /hV/ syllable they heard and rated its "goodness of fit." Results indicated that, in general, AE vowels were assimilated to the Japanese vowel category closest in articulatory vowel space. However, intrinsic duration information was utilized much more consistently when the syllables were embedded in a carrier sentence. That is, intrinsically long (or diphthongized) AE vowels were assimilated to long-vowel or two-vowel response categories more often in the sentence condition. Acoustical analysis indicated that some, but not all, of the variation in assimilation patterns could be explained by *F1/F2* variation. Implications for Japanese L2 learners of English are discussed. [Work supported by NIDCD.]

4aSC14. Response complexity effects on vocal reaction time in normal aged speakers. Michael S. Fozo and Ben C. Watson (Dept. of Otolaryngol., Rm. 171 Munger Bldg., New York Medical College, Valhalla, NY 10595)

Reaction time (RT) studies reveal that vocal responses present an exception to the pervasive finding of slowed performance in aged adults. Previous vocal RT studies that compared young and aged speakers elicited a single-word response. Inconsistent findings may be attributed to their use of a limited range of response complexity. Young speakers show a significant increase in vocal RT from vowel to sentence responses [Watson *et al.*, J. Voice 5, 18–28 (1991)]. This study tested the hypothesis that aged speakers in good physiologic condition show significantly longer vocal RT values relative to young speakers as responses increase in complexity. Vocal RT was recorded for isolated vowel, word, and sentence responses. The simple RT paradigm included variable foreperiods and intertrial intervals. Young subjects ranged from 24–31 yr. Aged subjects ranged from 68–85 yr. Group, response, and group x response effects were significant. *Post hoc* comparisons revealed significant between-group vocal RT differences for the word and sentence responses. Consideration of peripheral and central factors that may underlie the response complexity effect on vocal RT in aged speakers supports the conclusion that the origin of the complexity effect resides in central linguistic and/or motor planning for the response.

4aSC15. The effects of the attenuation of second and third formant frequencies on the recognition of stop consonant vowel syllables in aphasic and nonaphasic subjects. Lyn Goldberg (American Speech-Language-Hearing Association, 10801 Rockville Pike, Rockville, MD 20852)

The effects of the attenuation of second and third formant frequencies on the recognition of stop consonant vowel syllables (/p,b,t,d,k,g/ paired with the vowel /a/) in fluent and nonfluent aphasic and nonaphasic normal subjects were examined. Statistical analyses revealed significant differences between groups, between formant conditions, and between stop consonants. All groups responded similarly under the formant conditions, but the attenuation of *F2* appeared to account for the most striking disparity between consonant recognition scores irrespective of group. A series of confusion matrices showed that confusions made by aphasic subjects involved both place of articulation and voicing errors, particularly for nonfluent aphasics, and involved the substitution of more anterior consonants. Important cues for the voicing of anterior consonants appeared to be carried by *F2*. The finding that more intact speech production abilities appeared to facilitate more accurate speech recognition was interpreted as support for the motor theory of speech perception.

4aSC16. Lexical stress perception by hearing-impaired listeners. Dragana Barac-Cikoja and Sally Revoile (Ctr. for Auditory and Speech Sciences, Gallaudet Univ., Washington, DC 20002)

Identification of lexical stress in VCVs isolated from sentences was studied for moderately to profoundly hearing-impaired listeners. Two factors, sentential stress position and speech rate, were manipulated separately in the production of "You put VCV to bed." When sentential stress was manipulated, each of: you, put, VCV, bed, was stressed. When speech rate (fast and slow) was manipulated, the VCVs were from sentences with stress on the VCV only. Twenty VCVs for each speech rate and ten VCVs for each sentential stress position were tested. Hearing-impaired (n=33) and normal-hearing listeners (n=8) identified whether the VC or the CV was stressed. In contrast to the normal-hearing listeners, lexical stress

identification by the hearing-impaired listeners was significantly reduced for the fast speech rate. Both listener groups showed the same effect of sentential stress location: Performance was best when emphasis was on the VCV, and worst when either constituent adjacent to the VCV was emphasized. Results will be discussed with reference to the acoustic consequences of the production constraints and the accessibility of the prosodic cues to the hearing-impaired subjects as predicted by several audiological and psychoacoustic variables. [Work supported by NIH and the Gallaudet Research Institute.]

4aSC17. Comparison of Caseygrams for normal and pathological speech. P. G. Vaidya (School of Mech. and Mater. Eng., Washington State Univ., Pullman, WA 99164), Alison Behrman (Hunter College, CUNY, New York, NY), C. R. Winkel (Washington State Univ., Pullman, WA 99164), and Sally Alforde (Latah Health Services, Moscow, ID 83843)

In this paper, vowel sounds produced by healthy speakers and those produced by speakers suffering from certain pathologies are compared using differential trans-spectrograms, or the "Caseygrams." These Caseygrams are generated as follows: First, the vowel sound samples are optimally reconstructed to reduce leakage errors. The data are then partitioned into many smaller samples. Each of these is transformed into the frequency domain. These FFTs are used to calculate two different types of trans-spectral coherences (TSCs). [Please see Vaidya and Anderson, J. Acoust. Soc. Am. **89**, 2370–2378 (1991).] These two TSCs are specifically developed to measure the interaction of the main harmonics with the subharmonics. Positive results in this differential trans-spectrogram (Caseygram) indicate that chaos is present, either in a nascent state or fully developed state. It is expected that these Caseygrams would give an early warning of impending disorders and would also help monitor recovery from disorders such as damage to the speech center of the brain following a stroke.

4aSC18. Optimization for source waveform synthesis of pathological voices. Brian C. Gabelman, Jody Kreiman, Bruce Gerratt (Voice Lab., Div. of Head/Neck Surgery, UCLA, 31-24 Rehab. Ctr., Los Angeles, CA 90095-1794), and Abeer Alwan (UCLA, Los Angeles, CA 90095-1794)

This paper describes an algorithm that fits analytical source models of the glottal volume velocity to glottal source waveforms from inverse-filtered vowels. The analytical models include the KGLOTT88, LF, and simplified LF [Qi and Bi, J. Acoust. Soc. Am. **96**, 1182–1185 (1994)]. In the first stage of the algorithm, curve features (peak values, zero crossing, period, etc.) are automatically extracted from samples of pathological voices and least-squares fitting is performed. The parameters are then fine-tuned based on the perceptual quality of the synthesized voices. Goodness of approximation of the resultant source function is determined by perceptually comparing the final synthesized vowel sounds to the original sound. Limitations of current source models with respect to modeling pathological voice will be discussed. [Work supported by NIDCD.]

4aSC19. Acoustic characteristics of voice source for pathological voices: Some case studies. T. V. Ananthapadmanabha (Voice and Speech Systems, 53, Temple St., Malleswaram, Bangalore 560003, India), B. Mallikarjun, and Dipti Mallikarjun (The Pears Speech & Hearing Ctr., Ellisbridge, Pritamnagar, Ahmedabad 380 006, India)

The aim of this study is to examine the voice source signal of pathological voices. Pathological cases of vocal nodules, polyps, paralysis, malignancy, and laryngitis have been investigated. Some pre- and postoperative cases have also been studied. The sustained vowel /a/ of the subject has been recorded. The acoustic signal has been analyzed using an inverse filtering technique to obtain the voice source signal (derivative of glottal air flow). The voice source signal is displayed and marked for detailed examination. Acoustic recording, analysis, and marking is done using a VAGHMI system. For each type of pathology, certain consistent features deviating significantly from the normal cases have been noted in the voice source signal. For example, occurrence of double peak during the open phase, prolongation of the interval near glottal air flow peak, absence of clearly defined negative peak near epoch, etc. Interpretation of the voice source signal in terms of the vocal fold vibratory pattern and pathology has been attempted. Such an interpretation is validated against direct laryngoscopic observation where ever available. This on-going study suggests the possibility of differential diagnosis of the vocal fold pathology by acoustic analysis of vowel sound.

4aSC20. Analysis of "cold-affected" speech for inclusion in speaker recognition systems. Renetta G. Tull (Communications Sciences and Disorders Dept., Northwestern Univ., Evanston, IL) and Janet C. Rutledge (Northwestern Univ., Evanston, IL 60208-3118)

Parameters of speech produced during winter weather are analyzed to determine the potential influence of distorted voices due to common cold on automatic speaker recognition systems. This research examines speech recorded before, during, and after bouts with the common cold. "Cold-affected" speech shows noticeable acoustic and phonetic differences between test sessions. Examining the different states of one male speaker's voice allows an opportunity to address the speaker intersession problem that exists in speaker recognition technology. Specific emphasis is placed on the contrasts in various sessions of TIMIT inspired sentence "She had your dark suit in greasy wash water all year." It also analyzes phonetic contrasts and looks at differences in formant patterns. Phonetic transcriptions of "cold" and "normal" sessions reflect changes in place of articulation. Perceptual and acoustic analyses reveal pauses and epenthetic syllables that are not constant throughout all sessions. "Cold-speech" also produces noise caused by hoarseness and coughing. Research presented here shows evidence of variability beyond "normal" speech patterns. Contributing to the information on intraspeaker variability, this work provides a preliminary analysis of "cold-affected" speech in an effort to include "abnormal" speech and speech affected by communications disorders into the framework of automatic speaker recognition.

4aSC21. Analysis of the glottal excitation of intoxicated versus sober speech: A first report. Kathleen E. Cummings (Digital Signal Processing Lab., School of ECE, Georgia Tech, Atlanta, GA 30332-0250), Steven B. Chin, and David B. Pisoni (Indiana Univ., Bloomington, IN 47405-1301)

The objective of the research reported here was to perform an analysis of the voicing characteristics of sober and intoxicated speech in order to assess the possibility of detecting intoxication from the speech signal. Because the glottal excitation is the result of complicated and intricate motions by the vocal folds, it should be significantly affected by alcohol use. In this study, excitation parameters were extracted from non-nasalized vowels in eight utterances spoken in both sober and intoxicated conditions by four speakers. Fifteen parameters, including pitch, pitch contour, rms intensity measures, and measures of shimmer and jitter, were extracted directly from the speech waveform. As expected, the most significant differences between sober and intoxicated speech were in parameters measuring perturbations in adjacent pitch periods. Also as expected, there were no significant differences in more global parameters such as average pitch. Additionally, glottal waveforms were extracted from two of the speakers in both conditions using an adaptation of Wong's closed-phase glottal inverse-filtering method. Again, the most significant differences between sober and intoxicated glottal excitation were found in pitch-period-to-pitch-period variability. [Supported by a grant from the Alcoholic Beverage Medical Research Foundation to Indiana University.]

4aSC22. Speech motor coordination in children with phonological disorders. Marios Fourakis, Ying Xu, and Jan Edwards (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210)

This study examined the question of whether children with phonological disorders have problems in speech motor coordination. Four children with phonological disorders, four typically developing children matched for age and gender, and four adults were asked to say multiple repetitions of "bad dog" at normal and fast rates of speech. The productions were digitized and formant tracks were obtained using LPC analysis. Standard deviations were calculated over the second-formant values frame by frame along the four 50-ms spans of the CV and VC transitions. The $F2$ tracks of the adults were much less variable than those of both groups of children. The $F2$ tracks of the phonologically disordered children were more variable than those of their typical age peers for three of the four transitions; however, the $F2$ tracks of the phonologically disordered children were less variable than their age peers for the initial "ba" transition, the only transition in which the tongue was not involved in both the consonant and the vowel articulation. This result suggests that children with phonological disorders may have difficulty in rapidly sequencing gestures that involve conflicting configurations for the same articulator.

4aSC23. A numerical measure of alcohol depression. Harb. S. Hayre (Impairment Measures, Inc., 10405 Town and Country Way #203, Houston, TX 77024-1100)

Consumption of alcoholic beverage is said to cause depression which is then followed by a high. A literature review shows that this depression had not been measured numerically. Chemical impairment measure (CMI) based on speech was used by the author to detect and measure the physiological/neurological depression. CMI ranges from zero to three on a special scale. Twenty subjects were given a double drink of whiskey each, and their CMI were measured at 2 min intervals starting 2 min prior to alcohol ingestion and continued for 60 min thereafter. CMI, in all cases was observed to initially decrease to a minimum, then monotonically increase to a maximum, and eventually decrease thereafter until it reaches a value near the predrink value. The time to reach the minimum as well as the depth thereof varied from one person to the next verifying the fact that each person has their own tolerance level.

4aSC24. Effects of syllable duration on stop-glide perception by humans and monkeys. Joan M. Sinnott, Charles H. Brown, and Kelly W. Mosteller (Comparative Hearing Lab., Psych. Dept., Univ. of South Alabama, Mobile, AL 36688)

Humans and monkeys were compared in their perception of phoneme boundary shifts along two synthetic stop-glide /bα-wα/ continua differing in overall syllable duration (150 vs 320 ms). Humans were first tested with a written identification procedure and showed a boundary shift to longer transition durations with increased syllable duration, as previously reported in the literature. Humans and monkeys were then tested with a low-uncertainty discrimination procedure but showed little evidence of a sensory-level discontinuity underlying the identified boundaries: Instead sensitivity tended to follow Weber's law. Finally, both humans and monkeys were tested with a *go/no-go* identification procedure specifically designed for monkeys. Similar stop-glide boundaries emerged and shifted with increased syllable duration for both species, indicating that monkeys make good models of human stop-glide sensitivity in identification procedures that involve higher level attentional and memorial processes. [Work supported by NIDCD.]

4aSC25. The McGurk effect for non-native speech sounds perceived as nonspeech. Lawrence Brancazio (Haskins Labs., 270 Crown St., New Haven, CT 06511 and Univ. of Connecticut)

The McGurk effect—the integration of visual information into speech perception—is well documented for English consonants. However, a recent attempt to extend the effect to musical events (violin plucks and bows) showed only a weak visual influence on perception [Saldaña and Rosenblum, Percept. Psychophys. **54**, 406–416 (1993)]. Thus the conditions required to produce a strong McGurk effect are unknown. The effect may be specific to perception of native phonological categories; alternatively, it may apply to dynamic vocal-tract events in general. This study tested for a McGurk effect with click consonants, which are phonological in some African languages, but are treated as nonspeech by English speakers [Best, McRoberts, and Sithole, JEP:HPP **14**, 345–360 (1988)]. Audio bilabial, dental, and lateral clicks, spoken in isolation and coarticulated with a vowel, were dubbed onto video presentations of each click. Parallel stimuli were made from English stops in syllables and with their excised bursts only. English-speaking subjects made bimodal matching judgments on the dubbed stimuli. Results revealed a visual influence on perception of the non-native sounds that was somewhat weaker than for native speech, although stronger than Saldaña and Rosenblum's effect for violin sounds. [Work supported by NIH.]

4aSC26. Speakers' production of word frequency according to semantic and pragmatic contexts and to a production versus perception representation. Jan Charles-Luce, Kristin R. Dolena, Lorrie Chappell, and Kelly M. Dressler (Language Production Lab., Dept. of Communicative Disorders and Sciences, Park Hall, Univ. at Buffalo, Buffalo, NY 14260)

It has been shown that speakers adjust their articulation according to the amount of semantically biasing information in the whole utterance. Furthermore, subjects modify their speech production for the benefit of a listener. In the present investigation, speakers' production of high- and low-frequency words was examined when produced in two semantic contexts (semantically biasing and semantically neutral) and in two pragmatic contexts (presence or absence of an overt listener). In addition, speakers were instructed to produce the target words in a production mode versus a perception mode of experience. For example, speakers were instructed to interpret the words according to how frequently they "say" the word or how frequently they "hear" the word. The results of acoustic measurements showed that subjects' production of high- versus low-frequency words was affected by the semantic and pragmatic contexts. However, subjects modified their articulation differently depending on the mode of experience in which they interpreted the words' frequency. These results will be discussed according to pragmatic compensation and semantic activation in speech production and, in particular, how speech production may be affected by the speakers' representation of lexical items according to their production versus perception mode of experience. [Work supported by NIDCD.]

4aSC27. A study of lecture and computer-based teaching of articulatory and acoustic aspects of American English vowels. William D. Clarke III (Dept. of Communication Sciences and Disorders, Univ. of S. Florida, Tampa, FL)

Students in introductory courses on speech science have considerable difficulty in understanding the articulatory and acoustic phonetics involved in the production of vowels. A computer-based course of instruction has been developed that is intended to increase students' understanding of the interrelationships of the important concepts in this difficult area [Jackson *et al.*, J. Acoust. Soc. Am. **95**, 3014(A) (1994)]. The present experiment evaluated the computer-based exercises along with the traditional lecture method. Students were assigned to two groups, and each group received both the lecture and computer instruction but in a different order. Cognitive structures tests using similarity ratings were administered to assess students' understanding of the material before and after each type of instruction. Twenty concepts from the disciplines of articulatory and acoustic phonetics of vowels were paired for similarity ratings. Students' mental models were compared to those of experts using "Pathfinder" network, multidimensional scaling, and profile correlations. The students' shift toward expert-like patterns was small after lecture-only or computer-only presentations. The shift was moderate when the computer was first and lecture was second. The shift was large when the lecture was followed by the computer.

4aSC28. Perceptions of music and speech based on the autocorrelation and interaural cross-correlation functions. Yoichi Ando (Graduate School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe 657, Japan)

Based upon a model of auditory pathways which consists of the autocorrelation and the interaural cross-correlation mechanisms [Ando, *Concert Hall Acoustics* (Springer-Verlag, New York, 1985)], perceptions of music and speech as well as subjective responses of sound field in rooms are well described. This paper reviews some subjective attributes from recent investigations. The factors extracted from the autocorrelation function (ACF) of source signals include: (1) the energy of sound signal; (2) the effective duration of the normalized ACF; and (3) peaks as a function of its delay. These factors are deeply related to (1) loudness, (2) subjective attributes of the temporal structure of sound field, (3) the missing fundamental of complex sounds, respectively. Factors which may be extracted from the interaural cross-correlation function are included: (1) the IACC, the maximum value in the range of the interaural delay time; (2) τ_{IACC}, the delay of the IACC; (3) W_{IACC}, the width of IACC. The first factor well describes the clarity of speech as well as subjective preference of sound field. The second factor may be related to the apparent source width (ASW), and the last one is related to the image shift of sound source.

4aSC29. Infant preference for harmonic closure in musical sequences. Michael L. Tucker, J. David Smith, and Peter W. Jusczyk (Dept. of Psych., SUNY, Park Hall, Buffalo, NY 14260)

Previous investigations of Western tonal sensitivity suggest that young infants may have rudimentary tonal knowledge, for they notice altered notes and chords in a variety of change/no-change discrimination tasks. In contrast, infants' preference for different tonal structures was evaluated here using the head-turn preference procedure. A group of infant subjects (age 6 months) was presented diatonic melodic sequences, identical in duration, tempo, and intervallic distance, that either reached harmonic closure (closed) or did not (open). Six closed–open pairs were constructed by matching sequences for pitch contour. The sequences were presented in a random order, and orientation times to each sequence were recorded. Surprisingly, 6-month-olds revealed a significant preference for the harmonically open melodic sequences. An additional study examines whether 9-month-olds exhibit a reversal of this preference, or whether harmonic incompleteness continues to recruit attentional processes. These findings are considered in light of the child's growing experience with Western tonal music, and auditory-perceptual reorganizations that may occur during the latter half of the first year in music and speech. [Work supported by NIDCD.]

4aSC30. Gestural overlap analysis of lenis stop reduction in Korean. Sook-hyang Lee (Dept. of English Language and Literature, Wonkwang Univ., Iksan, Korea 570-749)

Korean lenis stops have been reported to become voiced intervocalically (Kagaya, 1974) and are often reduced to sonorants in casual speech with different degrees of reduction depending on the place of articulation (Lee, 1995). Velar lenis stops are more often reduced than the labials and coronals, which makes sense from the point of view of articulatory phonology (Browman and Goldstein, 1990). The oral gesture for a velar stop is on the same vocal tract tier (tongue body tier) as the neighboring vowels, so it should be more affected by overlap and blending with the vowel gestures than the closure gestures for other stops. This study investigates in what vowel environments lenis stop closure shows the most reduction. /VCV/ tokens in carrier sentences were recorded and acoustic analysis was done. The results showed that generally, reducing lenis stop closure is most in the environment of preceding and following low vowels, and least in the environment of high vowels, where the tongue body gestures of neighboring vowels are more compatible with the stop's gesture. Labials, especially when they are in the high rounded vowel environment, showed a larger amount of reduction, which makes sense from the point of view of articulatory phonology, since one of the oral gestures for a labial stop is on the same vocal tract tier, lip tier, as the surrounding rounded vowels.

THURSDAY MORNING, 16 MAY 1996 REGENCY D, 8:00 TO 11:45 A.M.

Session 4aUW

Underwater Acoustics: Visualization of Acoustic Propagation Mechanisms

Shira L. Broschat, Chair

School of Electrical Engineering and Computer Science, Washington State University, Pullman, Washington 99164-2752

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Criteria and examples of the use of visualization in underwater acoustics. R. A. Zingarelli and S. A. Chin-Bing (Naval Res. Lab., Acoust. Simulation Section, Stennis Space Center, MS 39529)

Visualization in underwater acoustics is a relatively new field, with a peculiar set of promises and pitfalls. Advanced computer graphics can provide insight into physical processes, but only if the tools are simple and flexible enough to provide 'on the fly' visualization as the research is being done. Furthermore, 'reality checks' against standard benchmark solutions must be performed at every opportunity. Here, examples are presented of utilitarian visualization for time-domain sound propagation, how these are checked against reference solutions, as well as some interesting cases where these checks were not made. [Work supported by ONR.]

8:30

4aUW2. Bottom interactions of pulse propagation and corresponding real-time reverberation. Kevin B. Smith (Dept. of Phys., Code PH/Sk, Naval Postgrad. School, Monterey, CA 93943)

A simple, memory-intensive algorithm using a split-step Fourier cw PE model to compute the time-domain animation of a pulse has been developed. By examining such an animation in the presence of a rough, deep ocean bottom, the significance of secondary multipaths on predicted reverberation returns was found to be non-trivial. The existence and relative magnitude of these secondary arrivals has been confirmed in measured data. An additional algorithm has recently been developed to compute the exact two-way travel times associated with bottom interface reverberation by convolving the one-way travel times from the source and the receiver. Presented here is a simultaneous visualization of the one-way pulse propagation and the corresponding two-way reverberation. To determine the influence of the secondary multipaths on reverberation, the bottom has been altered in the near field to remove the multipath influence. This also shows the errors encountered in predicting reverberation when only the incident field is computed. [Work supported by ONR.]

8:55

4aUW3. Animations of acoustical pulse (time-domain) signal propagation in a realistic waveguide to high frequencies. Nataly A. Sidorovskaia (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148) and Michael F. Werby (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529)

Recent developments have made it possible to perform propagation calculations in realistic oceans to high frequencies. It is possible to deal with on the order of hundreds of modes rather quickly and by storing the frequency-domain field over some specified range over the water column and for a fine frequency mesh, it is possible by a new Fourier transform algorithm to reconstruct time-domain solutions in reasonable times. Time frames can be produced for various pulse forms and realistic animations can be created. Thus this enables concentration on the physics of pulse arrivals as well as distortion effects from submerged inclusions. The insight that arises from these calculations, whether they are still shots or actual animations, is very revealing and affords an additional perspective of the complete process that otherwise would not be revealed by less ambitious calculations. A variety of animations and still shots for several physical environments are presented and much of the theoretical results with the intuitive powers that only animations offer are described. [Work supported by Naval Research Laboratory and Office of Naval Research.]

9:20

4aUW4. Interface waves in seafloor propagation and scattering problems. Ralph A. Stephen and S. Thompson Bolmer (Woods Hole Oceanograph. Inst., 360 Woods Hole Rd., Woods Hole, MA 02543)

Interface waves often play a significant role in seafloor propagation and scattering problems. Because they are not solutions to the pure acoustic wave equations, but require a nonzero shear modulus, they are frequently overlooked in model and inversion studies. There are two areas in particular where interface waves are relevant in underwater acoustics: (1) when the source and receiver are close (in terms of wavelengths) to the seafloor, and (2) when interface roughness or heterogeneities near the interface act as secondary scatterers. The first case is important for low-frequency propagation in shallow water. The second case is important in scattering problems such as low-angle, monostatic backscatter or the scattered field of seafloor or buried ordinance. Insight into the generation and propagation mechanisms of interface waves can be gained through time-domain shapshots of the wavefield and by analysis of time series record sections. Animations of wavefront propagation in vertical slices through the seafloor show clearly the energy partitioning between the various body and interface wave types. [Work supported by ONR.]

9:45

4aUW5. Understanding acoustic propagation in shallow water via animations. John B. Schneider, Patrick J. Flynn, and Shira L. Broschat (School of Elec. Eng. and Comp. Sci., Washington State Univ., Pullman, WA 99164-2752)

Modeling the propagation of acoustic energy in shallow water is difficult due to the complexity of the environment. Roughness at the air–sea interface and bottom, inhomogeneities in the water column and sediment, and discrete scatterers affect the propagation. Although many modeling techniques are available to study this problem, arguably none is general enough to provide robust and accurate results over a large region when several complicating physical features are present. Instead, simplifying assumptions are often made, and the effect of such assumptions on the model's validity is uncertain. Here, propagation in shallow water is studied via animations of the results obtained using the finite-difference time-domain method (a powerful, but computationally expensive, modeling technique with few inherent assumptions). Such animations provide insight into the relative importance of certain physical features that is not easily obtained from "still" data. Two different types of animations are discussed. One employs a "side-by-side" comparison in which time-domain propagation is shown in both the presence and absence of a certain physical feature. In the other, time is held constant and a physical feature is varied from frame to frame. Creation of animations using freely available software is also discussed. [Work supported by ONR.]

10:10–10:30 Break

10:30

4aUW6. Modeling target strength of reflective objects using computer ray tracing technique. Gerard P. Gay (Dept. of Mech. Eng., Cooper Union, 51 Astor Pl., New York, NY 10003) and Daniel R. Raichel (Acoust. Res. Ctr., Cooper Union, New York, NY 10003)

Monostatic target strength of perfectly reflective geometric objects may be predicted through the use of a non-recursive ray tracing technique. This solution is based on the analytical infinite series solution of the wave equation. The ray tracing theory embodies the premise that the properties of sound, with wavelengths small compared to the size of the target objects, are analogous to those entailed in the propagation of electromagnetic waves. Simple geometric objects including spheres, disks, and cylinders are generated as object-oriented three-dimensional models; objects are ray traced so the model may be viewed and results may be calculated. The data produced by the use of this technique agree with theoretical results, thereby suggesting the validity of this method.

10:45

4aUW7. Modal time series structure in range and depth of a shallow-water area. David P. Knobles (Appl. Res. Labs., Univ. of Texas, P. O. Box 8029, Austin, TX 78713-8029)

A theoretical analysis of pressure time series generated by small explosive sources and recorded on both an HLA and a VLA deployed in the Hudson canyon region off the New Jersey coast near the AMCOR 6010 borehole is presented. The SVP in the water column has an isovelocity layer down to a depth of about 20 m followed by a strong negative gradient to about 45 m where the profile becomes approximately isovelocity to theseafloor at 73 m. This sound-speed structure creates a unique time series structure on the VLA as a function of source range. The data are simulated with a broadband normal mode- approach recently discussed in the literature [J. Acoust. Soc. Am. **98**, 1682–1698 (1995); IEEE J. Ocean. Eng. **21** (1) (1996)]. The representation of the spatial and temporal structure of the time series in terms of a modal structure reveals several unique aspects of the structure of the SVP in the water column and the geoacoustic structure of the bottom. The details of the modal structure of the sound field clearly define the propagation mechanisms in the water column and the interaction with the seafloor and sub-bottom sediment layers. [Work supported by Advanced Research Projects Agency.]

11:00

4aUW8. Mode coupling and mode stripping in propagation across the range-dependent Atlantic Generating Station (AGS) site. Indra Jaya (Rensselaer Polytechnic Inst., Troy, NY 12180), Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

The complicated patterns of modal structure and attenuation observed in sound propagation through a range-dependent shallow-water environment are used to investigate and visualize the mechanisms of mode coupling and stripping. Broadband experiments have been carried out at the geologically known AGS site in the past few years. This presentation reports on a recent experiment which provides significantly more acoustic data along different propagation tracks and, consequently, permits high resolution of physical processes in range. A mode-filtering technique is used to construct modal structure from the data, while modal attenuation is obtained from spectral ratios. The evolution of modal structure is determined as a function of frequency and range. This is examined in view of environmental parameters along a selected propagation track. Numerical-simulations are carried out using interpolations of the known geoacoustic environment and employing the KRAKEN normal mode and RAM parabolic equation models. Results from these simulations are compared with the experimental data.

11:15

4aUW9. Mapping acoustic echosounder data to human color vision. Frank A. Boyle (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A colormapping algorithm was developed and applied to echosounder data from the seafloor. The technique offers a simple method of conveying subtle information about the seabed that might be useful in sediment classification. Features with different spectral content appear in a processed image with different shades of color. Broadband scatterers, such as rigid objects or interface roughness emerge as white or gray features, while narrow-band scatterers, such as trapped gas bubbles appear with a hue corresponding to their resonance frequency. The method has the potential to resolve the ambiguity between rigid and gassy sediments, which are indistinguishable in a grayscale display. A description of the colormapping algorithm, followed by a presentation of the results of its application to echosounder data and a discussion of their interpretation is given. [Work supported by NRL.]

11:30

4aUW10. Wave field visualization as an interpretational tool in borehole geophysics. V. N. Rama Rao, Henrik Schmidt, and Kim Vandiver (Rm. 5-007, MIT, Dept. of Ocean Eng., 77 Massachusetts Ave., Cambridge, MA 02139)

A hybrid model for computing the radiation from a borehole while drilling has been developed. A model for modal propagation in a radially layered borehole is coupled with the OASES code for full wave modeling of the resultant seismic wave field. The coupling is performed using the concept of effective sources. First, an equivalent seismic source array that produces the same far-field radiation as the borehole is determined. This source array is then introduced into the layered earth model and the resulting seismic field is computed. Three propagating modes exist in the borehole in the presence of the drill-pipe. The individual modes radiate into the formation in a pattern that is dependent on geophysical properties, with the resulting radiation pattern being extremely complex and difficult to interpret physically. Here, visualization and animation in particular has proven to be an extremely useful tool. For example, animation allows the direct identification of the complex arrival pattern recorded on surface mounted geophones. The fundamental differences between the radiation patterns in fast and slow formations are also clearly demonstrated using computer visualization and animation.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

U.S. Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protection

D. L. Johnson, Chair S12
EG&G Special Projects, Albuquerque Operations, Albuquerque, New Mexico 87119-9024

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
U. S. CERL, P.O. Box 4005, Champaign, Illinois 61820

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

E. H. Berger, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 94/SC12, Hearing Protection
Cabot Safety Corporation, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Standards Committee S12 on Noise. Working group chairs will report on their progress for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will also be discussed, with reference to the international standards under preparation. A report will be given on the meeting of ISO/TC 43/SC1 in Pretoria, South Africa, held in February 1996. The Chairs of the respective U.S. Technical Advisory Groups (H. E. von Gierke and E. H. Berger) will report on current activities of these international Technical Subcommittees under ISO.

Scope of S12. Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation, and control; including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.

THURSDAY AFTERNOON, 16 MAY 1996 MT. RUSHMORE, 1:30 TO 4:45 P.M.

Session 4pAA

Architectural Acoustics and Noise: Current Topics in HVAC Noise

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, Connecticut 06066

Timothy J. Foulkes, Cochair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—1:30

Invited Papers

1:40

4pAA1. New developments in noise control for HVAC systems. Timothy J. Foulkes (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776)

The HVAC equipment industry is not known for rapid change. Although control systems have improved rapidly over the past few years, fans are still much the same as they were 20 years ago. Accessories such as air diffusers and vibration isolators have gone through small, incremental changes in the same time period. In the past 12 months, several new products have caught the author's attention. These represent significant improvements in noise control and/or design flexibility and installation quality. This presentation will include product information on these new products along with a discussion of their pertinent advantages. Industry trends such as wider availability of sound data and custom engineered air handling units will also be discussed.

2:05

4pAA2. Assessment of HVAC sound power data for sensitive spaces. Kevin C. Miller and Martin J. Beam (Miller Henning Assoc., Inc., 6731 Whittier Ave., Ste. A110, McLean, VA 22101)

Certification testing of air handling unit sound power (PWL) indicates that individual unit PWL can vary significantly from manufacturer published data. Published data are typically based on a limited number of actual tests with results extrapolated for other fan sizes and operating conditions. Although published data are normally acceptable for routine applications, indiscriminate use for sensitive designs (studios, theaters, conference centers, etc.) can result in excessive finished space noise levels. The fact that the design goals have been exceeded can be accurately documented. The exact reason for the exceedance, however, cannot be as firmly established in a complex system. Certification PWL testing of air handling units prior to installation can detect PWL variations from the published data. Individual unit PWL certification was often not cost effective prior to the developement of sound intensity and the establishment and use of recognized standards for *in situ* PWL testing (e.g., ASA 104-1992, ANSI S.12.12.1992). Case histories to be presented document fan PWL variations from published data of up to 10 dB and higher in certain octave bands.

2:30

4pAA3. Air pedestal diffusion method of introducing air silently to spaces. Robert Essert and Damian Doria (Artec Consultants Inc., 114 W. 26 St., Floor 9, New York, NY 10001)

In recent years the drive to air condition performance halls with greater energy efficiency and comfort has brought supply air terminals closer to the ears of audience members. Introduction to North America of the European approach of air volume displacement has required development of silent air terminals that coordinate with audience seating and allow greater efficiency and comfort. Artec Consultants Inc. has worked with leading seating manufacturers to develop pedestal chair supports and other floor mounted devices that function as terminals for air supplied from underfloor plena. The Ford Centre in North York Ontario incorporates one such design, as will several additional facilities in North America and Europe that are presently under construction. Completed facilities have met the design noise levels. This paper will examine the recent development of these terminal devices and discuss the benefits and complications associated with their implementation.

2:55–3:10 Break

3:10

4pAA4. Duct breakout sound associated with ducts constructed of fiberglass duct board. Douglas D. Reynolds and Keith M. Degner (Ctr. for Mech. & Environ. Systems Technol., College of Eng., Univ. of Nevada Las Vegas, Las Vegas, NV 89154-4040)

Sound breakout from rectangular HVAC ducts constructed of fiberglass duct board has been investigated. Duct sizes that were tested were 12×12, 12×24, 12×36, 24×36, and 36×36. All dimensions are in inches. Four duct lengths for each duct size were tested: 4, 8, 12, and 16 ft. The sound breakout values for the fiberglass duct board were compared to corresponding values obtained from similar sized rectangular sheet metal ducts. Major objectives of the study were to obtain duct transmission loss values for rectangular ducts constructed of fiberglass duct board, to compare these values with corresponding values obtained from similar sized rectangular sheet metal ducts, and to specify conditions in which rectangular ducts constructed of fiberglass duct board can be used in situations where the control of duct breakout noise is important.

3:35

4pAA5. The influence of fans on low-frequency noise and rumble in HVAC systems. Jerry G. Lilly (JGL Acoust., Inc., 6421 Lake Washington Blvd. N.E. #209, Kirkland, WA 98033)

Low-frequency noise and rumble problems can occur in HVAC systems due to problems with the fan, the duct sytem, the fan/duct interface, and even the control system. This discussion deals with fan related problems only. Items that will be discussed include: the importance of fan-type selection (centrifugal, axial, plenum, etc.), selecting the right fan size for the required duty, the importance of aerodynamic conditions at the fan inlet and outlet, and the problems associated with VAV control mechanisms. The discussion will also address the importance of proper vibration isolation and how to prevent common problems. Specific project examples will be included.

Contributed Papers

4:00

4pAA6. Computer program for conducting acoustical analyses of HVAC systems. Douglas D. Reynolds and Scott C. Mitchell (Ctr. for Mech. & Environ. Systems Technol., College of Eng., Univ. of Nevada Las Vegas, Las Vegas, NV 89154-4040)

An interactive Windows-based computer program has been developed that can be used to conduct complete acoustical analyses of HVAC systems. The program can be used to track sound from a HVAC sound source such as a fan to a room. Both sound from a single path between a sound source and room or from multiple paths between a sound source and room can be investigated. The program has a full set of editing features that include change, delete, and insert options which allow an individual to interactively design an HVAC system to meet specified acoustic design criteria. Design templates for systems in which multiple acoustical analyses will be conducted can be developed and stored to minimize the time associated with inputting data to the program. The program has options for printing out tabulated results and noise criteria and room criterion curves for inclusion in reports.

4:15

4pAA7. Higher-order modal reflection and transmission in acoustic waveguide step discontinuities. Ralph T. Muehleisen (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

Analytic expressions for the transmission and reflection coefficients of higher-order modes at a step discontinuity in an acoustic waveguide are developed. The equations take on the familiar form of an impedance discontinuity when written in matrix form. Numerical results show that there is significant modal coupling which indicates that the coupling cannot be ignored in the design of duct systems.

4:30

4pAA8. Higher-order modal reflection and transmission in acoustic waveguide junctions. Ralph T. Muehleisen (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

Analytic expressions for the transmission and reflection coefficients of higher-order modes in acoustic waveguides are developed and explored. When written in matrix form the equations take on the standard form of an impedance discontinuity. Numerical results for a right angle bend and a T junction show that there is significant coupling between modes, especially to the plane-wave mode. The results indicate that modal coupling cannot be ignored in the design of active and passive attenuators that need to work at higher frequencies.

THURSDAY AFTERNOON, 16 MAY 1996 CANYON HALL, 1:30 TO 5:20 P.M.

Session 4pAB

Animal Bioacoustics: Intra-Specific Variations in Animal Vocalizations II

Robert H. Benson, Chair

Center for Bioacoustics, MS 3367, Texas A&M University, College Station, Texas 77843-3367

Chair's Introduction—1:30

Contributed Papers

1:35

4pAB1. Quantitative analysis of corvid (class aves) vocalizations as a possible taxonomic tool. Mary K. Coldren (Ctr. for Bioacoustics, Texas A&M Univ., College Station, TX 77843-3367)

Vocalizations of scrub jays (*Aphelocoma coerulescens texana* and *A.c. woodhouseii*) and gray-breasted jays (*A. ultramarina couchii*) were recorded at eight sites in Texas. All three taxa were considered allopatric. While many call types were not consistent in structure and tended to intergrade, two structurally distinct call types were identified as shared by all taxa at all sites. Differences in acoustic structure of both call types was greatest between species. Acoustic structure between subspecies was less distinct, with unique acoustic structures occasionally being shared with the nearest population of another subspecies. Morphometric analysis revealed a similar pattern, with the most differentiation at the specific level, followed by subspecific and, lastly, by populational differentiation. Of particular interest was a small, isolated population of *A. coerulescens* found in the Texas panhandle. Although breeding populations may have existed for only two or three decades, morphometric analysis of both call types showed marked vocal differentiation from other Texas scrub jays. Analysis of one call type suggested separation from the other populations of scrub jays at the subspecific level, while the other call type suggested separation at the specific level.

1:50

4pAB2. Dialects in South Pacific humpback whale song. D. A. Helweg (Dept. of Psych., Univ. of Auckland, Private Bag 92019, Auckland, New Zealand), D. H. Cato (2DSTO, Pyrmont. NSW, Australia), P. F. Jenkins (Univ. of Auckland, Auckland New Zealand), and C. Garrigue (OR STOM, Noumea, New Caledonia)

Humpback whales (*Megaptera novaeangliae*) migrate annually between high-latitude summer feeding waters and low-latitude winter mating waters with a high degree of site fidelity. During migration and in winter waters, adult (male) humpbacks produce long complex songs. Song content is dynamic, but singers incorporate changes as they occur. Song sharing across regions is cultural, thus similarity of songs from different regions may be an index of reproductive isolation. Songs recorded in Tonga, New Caledonia, Eastern Australia, and Kaikoura, New Zealand, in the winter of 1994 are compared. The song from Kaikoura is the most southerly recording of song, over 2000 km south of winter migration termini. Seven themes were shared by all regions plus two themes not observed in Tonga. Differences in regional variants were most pronounced between Tongan and Eastern Australian song. New Caledonian and Kaikouran song were proportionately more like Eastern Australia. Kaikoura song was more similar to Eastern Australian song than Tongan song. These regional differences can be considered to be dialects that are maintained through cultural transmission of regional variants. The results suggest some migratory exchange among widely separate wintering regions of area V and speculation that song sharing occurs throughout migration and breeding seasons.

2:05

4pAB3. A nonparametric statistical approach to representation of variations within a class of marine mammal vocalizations. Thomas J. Hayward (U.S. Naval Res. Lab., Washington, DC 20375-5350)

A nonparametric statistical approach has been developed for representing variation within a class of vocalizations. This statistical representation permits computationally efficient *a posteriori* estimation, based on training in a supervised learning context, of the probability density associated with the class. This approach avoids the assumption of parametric forms for the probability density; parametric forms may not be suitable in cases where the probability density has a complex, multimodal character. Two applications of the statistical representation are described. First, the *a posteriori* densities associated with several classes of transients can be computationally "evaluated" on a new sample to provide a basis for classification of marine mammal vocalizations as well as a measure of confidence in that classification [J. Acoust. Soc. Am. **96**, 3312 (1994); **98**, 2969 (1995)]. Second, the statistical representation provides for quantitative characterization of variability using information-theoretic measures. Examples of both applications are presented. [Work supported by ONR and NRL.]

2:20

4pAB4. Interindividual variation in the songs of humpback whales. Adam S. Frankel (Bioacoustics Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850)

The function of humpback whale song remains elusive. Humpbacks primarily sing on the wintering grounds. Many hypotheses suggest a male advertisement function. It has been suggested that all whales of a population sing the same version of song, and its structure slowly evolves during and between singing seasons. Field observations of 1989 song indicated substantial variation in the song of humpbacks. The question remained, were the differences between individuals greater than those between successive songs of an individual. The songs of 11 whales were sampled during an 8 day period, to reduce the probability that song evolution would affect the results. Six units from three themes were sampled from four songs from each whale. Unit duration, lowest frequency, bandwidth, frequency of peak amplitude, and source level were measured. A repeated measures ANOVA found significant differences between individual whales and no such differences were found between successive songs of the same individual. These data indicate that there are real measurable differences between the songs of different individual whales. These individual differences could be used as a basis for female choice of males, based on song features. While this remains untested, these data are necessary for such a hypothesis.

2:35

4pAB5. Interspecific versus intraspecific variation in alarm call vocalization among two species of capuchin monkey. Kimberly S. Norris and Jeffery C. Norris (Marine Acoust. Lab., Ctr. for Bioacoustics, Texas A&M Univ., 5007 Ave. U, Galveston, TX 77551)

Inter- and intraspecific variation in alarm calls were compared for two species of free-ranging capuchin monkey: the wedge-capped capuchin (*Cebus olivaceus*) and the white-faced capuchin (*Cebus capuchinus*). Alarm calls in each case were made in the presence of the same predator, a boa constrictor. Vocalizations ($n=40$) were compared for commonly examined acoustic parameters (e.g., signal duration, frequency, and amplitude variables), as well as for vocal features found to be acoustic cues to the monkeys [e.g., bandwidth of dominant formant; change in frequency of the dominant formant (see Norris, 4aAB3)]. Additionally, correlations of spectrograms were performed using CANARY. Interspecific variability was found to be significantly greater than intraspecific variability for signal duration ($F=45.30; p<0.0001$), maximum frequency ($F=8.52; p<0.0059$), and formant number ($F=10.32; p<0.0027$). However, interspecific variability of calls was not significant for any of the acoustic parameters found to be used as perceptual cues for the two species. The evolutionary and ecological significance of these findings will be discussed.

2:50

4pAB6. Spectrogram contour features for intraspecific classification of narrow-band animal sounds. Ben Pinkowski (Dept. of Comput. Sci., Western Michigan Univ., Kalamazoo, MI 49008)

Recent research has indicated that contour features obtained from computer imaging are useful for characterizing some interspecific differences in narrow-band animal sounds [B. Pinkowski, J. Acoust. Soc. Am. **95**, 3419–3423 (1994)]. Contour features may also characterize the intraspecific differences of animal sounds that vary among individuals. To examine this possibility, six songs from each of ten eastern bluebirds (*Sialia sialis*) (5 males and 5 females) recorded during 1991 and 1992 in southwestern Michigan were digitized at 22 kHz, converted to spectrogram images, and analyzed for contour using Fourier descriptors (FD's). Following smoothing, thresholding, and segmentation, the spectrogram components corresponding to three or more basic notes were merged by a simple pixel aggregation algorithm to yield a single contour per song. The contour border was then used to compute a 512-point DFT. With 16 low-order FD's per song, 58 of the 60 songs (96.7%) were correctly classified by a linear discriminant function. Male songs were more varied and complex and hence more difficult to classify by contour alone than were female songs. [Work supported by NIH.]

3:05

4pAB7. Acoustic communication by imported fire ants. Robert Hickling, Wei Wei (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677), and Lavone Lambert (ARS Delta States Res. Ctr., Stoneville, MS 38776)

Although it is well known that ants generate sound by stridulation using gaster-flagging motion of the abdomen, acoustic communication by ants is generally believed to be almost nonexistent compared to chemical communication using pheromones [Holldabler and Wilson, *The Ants* (Harvard U. P., Cambridge, MA, 1990), pp. 255–259]. Contrary to this prevailing belief, it has been found that imported fire ants can use acoustic signals for meaningful communication. Signals have been recorded in situations such as alarm, distress, and attacking prey and also what appears to be an all's-well signal. These were analyzed and distinct time-modulation differences were found depending on the situation. Acoustic signals appear to be used for urgent communication. Ant response provides an indication of the significance of the signals. The frequency range is principally below about 2 kHz. The sound-pressure level of a general alarm signal was found to be approximately 40 dB (A), which is about three times the level of a faint whisper. Background noise inside an ant mound is low and interior tunnels can act as waveguides.

3:20–3:35 **Break**

3:35

4pAB8. Intraspecific variations in finback (*B. physalus*) 20-Hz doublets: An acoustic multipath interpretation. V. Premus and J. L. Spiesberger (Dept. of Meteorol. and Appl. Res. Lab., Penn State Univ., University Park, PA 16802)

Biologists have pointed to differences in received calls as evidence for some separation in finback populations in different locations [Watkins *et al.*, J. Acoust. Soc. Am. **82**, 1901–1912 (1987)]. More specifically, it has been concluded by Thompson *et al.* that the unique finback doublet interpulse intervals recorded in the Gulf of California constitute corroborating evidence for the hypothesis that "Gulf of California fin whales may be a separate population, distinct from fin whales in nearby Pacific regions" [Thompson *et al.*, J. Acoust. Soc. Am. **92**, 3051–3057 (1992)]. An alternative explanation is offered—that regional differences in finback doublet interpulse intervals may be accounted for by differences in acoustic propagation conditions. In this work, a broadband acoustic propagation code is combined with an empirically derived geoacoustic seafloor model for the purpose of interpreting finback doublet sequences recorded in the northern Gulf of California by Thompson *et al.* in 1987 [*ibid.*]. The result is that for ranges less than 30 km, it is possible to explain the appearance of the finback doublet in terms of multipath propagation through the water column and the seafloor. Further, the interpulse delay is demonstrated to be strongly range dependent. Scientists might consider the possibility that finbacks make use of multipath intervals to estimate distances to other finbacks, perhaps for reasons related to social behavior.

3:50

4pAB9. Removal of multipath effects from marine mammal vocalizations. Aaron M. Thode, Gerald L. D'Spain, William S. Hodgkiss, and W. A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093-0205)

Long-duration (>1 s) marine mammal vocalizations (i.e., blue whale calls) generate echos and other multipath effects [McDonald *et al.*, J. Acoust. Soc. Am. **98**, 712–721 (1995)], obscuring the original structure of the call, which may contain valuable anatomical information about the animal. Several deconvolution techniques are evaluated for their effectiveness in removing multipath effects from both a hypothetical and actual marine mammal call. The evaluation is first performed on a model of a whale call in an isovelocity waveguide with noise, where the source signal is known and can be compared with the output of the deconvolution schemes. The methods are then applied to two blue whale calls recorded during an experiment (SWellEx-3) in July, 1994 in 200-m water. The depth and range of the animal at those two times had previously been determined by matched field processing. As the position of the animal is known, the resulting multipath propagation can be computed and the accuracy of the techniques in identifying those multipaths determined.

4:05

4pAB10. An acoustic perception model for echolocation signals. Mathew J. Palakal, Uday Murthy (Dept. of Comput. Sci. IUPUI, 723 W. Michigan St., SL 280, Indianapolis, IN 46202), and Donald Wong (Indiana Univ., Indianapolis, IN 46202)

Bats that echolocate using biosonar pulses perceive their target as an auditory image by extracting acoustic cues conveyed in the target-reflected echoes. Neurophysiological studies suggest that a large majority of cortical neurons found in FM bats are delay sensitive. Moreover, these neurons are tonotopically organized and have a best delay (BD) at which they respond maximally to an input stimulus. In order to understand the role of these neurons during pulse-echo processing, a mathematical model capturing the functionality of delay sensitive neurons (DSN) has been developed. As a first approximation, response to a pulse-echo pair of a DSN is derived as, $y(t)=\exp[-i*(\beta-\delta)^2/2*\sigma^2]$, where β is best delay of the neuron, δ is arrival time of the echo, i is neural spikes, and σ is delay range of the neuron. A two-dimensional cortical response map (CORMAP) of DSNs is then developed based upon the best frequencies and the above stated response property with varying BDs. The CORMAP is then simulated using phantom targets containing multiple echoes. Once a pulse is sent and as the echoes begin to arrive for every δ_k, the CORMAP generates a response map as the DSNs fire. The firing patterns occur at various δ_k can be analyzed to estimate target distance and structural information. [Work supported by NSF.]

4:20

4pAB11. Experiments on the directional startle reflex in goldfish. Thomas N. Lewis, Theresa J. Woods, and Peter H. Rogers (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The acoustic startle reflex in goldfish (*Carassius auratus*) was studied experimentally in the large acoustic tank at Georgia Tech. Individual free swimming fish were placed in an acoustically transparent cage and positioned in the center of the tank. Their behavior was graded for a response to acoustic signals and the direction of response with respect to the sound source. A preliminary study indicated two interesting features. First, the fish was much more likely to respond in the "correct" direction (away from the source) when responding to subthreshold stimuli. Second, there was strong dependence of signal amplitude at threshold on fish size. For the two weight groups tested (1.6 g, $n=3$ and 23.6 g, $n=4$), the sound-pressure level at threshold was 15 dB lower for the smaller subjects. The initial study has recently been augmented with tests on a range of subjects weighing between these two extremes. Also, since only the first half-cycle of the acoustic stimulus is thought necessary to elicit the response, the concept of frequency to characterize the sound stimulus is inappropriate. Therefore, thresholds were measured as a function of signal rise time. A parallel effort has involved modeling a biologically relevant stimulus: the acoustic field generated by an attacking predator. [Work supported by ONR.]

4:35

4pAB12. Time-frequency analysis of potentially disturbing sounds in the environments of captive and wild fish. Joseph A. Clark (CDNSWC, Code 734, Bethesda, MD 20084 and COMB, Ste. 236 Columbus Ctr., 701 E. Pratt St., Baltimore, MD 21202), Jane A. Young (CDNSWC, Bethesda, MD 20084 and COMB, Baltimore, MD 21202), Amrit N. Bart, and Yonathan Zohar (COMB, Baltimore, MD 21202)

Time-frequency analysis provides a detailed means to characterize features of sounds which might affect animals. Graphic displays of the analyzed sound fields often reveal qualitative patterns and quantitative measures that can be used as a basis for comparing different sound fields. The displays can be animated and presented in conjunction with audio recordings of the sounds and with video recordings of the environments in which the sounds were produced in order to assist in identifying sources and effects of the sounds. Modular signal processing software, integrated with a high level programming language and operated on graphics workstations enable the researcher to select specific types of sounds which are found to occur in typical environments of selected animals. The system can also be used to synthesize calibrated sets of similar sounds to be used as inputs for controlled experiments investigating the effects of the sounds on animals. In this paper, a typical workstation-based system for performing time-frequency analysis of digitally recorded sound will be described. Operation of the system will be illustrated with results from analyses of sounds collected during a recent survey of the acoustic environments of captive and wild fish and with specialty sounds synthesized for current investigations of effects of noise on fish.

4:50

4pAB13. An "acoustic-signature" model of vowel evolution. Michael J. Owren (Dept. of Psych., Reed College, 3203 SE Woodstock Blvd., Portland, OR 97202)

Recent work in bioacoustics has emphasized the importance of cues to individual- and kinship-identity in "signature" calls produced by many animal species, including nonhuman primates. Applying the source-filter approach to sound production in monkeys and apes, it is proposed that low-pitched, tonal calls may be significantly better-suited to providing these indexical cues than are noisy or high-pitched vocalizations. As the supralaryngeal vocal tract is relatively inflexible in many primates, harmonically rich calls produced by different individuals should exhibit stable, subtly distinctive spectral characteristics due to intraspecies variation in vocal tract size, shape, and tissue properties. Based on field studies of calls from baboons and rhesus monkeys, it is suggested that protohominids routinely uttered vowel-like sounds long before the development of speech. Laboratory tests of pure-tone and formant frequency discrimination in monkeys and humans further indicates that detailed formant-related characteristics in these sounds were likely both functionally important and perceptually salient. Due to changes in facial morphology (probably reflecting dietary factors), shortening of the protohominid vocal tract created selection pressure for lower laryngeal position to maintain acoustic-signature cues. Laryngeal descent therefore set the stage for development of flexible vocal tract positioning, but was not itself an adaptation for speech.

5:05

4pAB14. Variation in whistle-type and rate produced by captive dolphins relative to in-water interactions with humans. T. G. Frohoff (Dolphin Data Base, 321 High School Rd. NE, Brainbridge Island, WA 98110), J. M. Packard, and R. H. Benson (Texas A&M Univ., College Station, TX 77843-2258)

The relative importance of expression and symbolic content in acoustic communication of cetaceans has been debated. These topics may be examined by observing how general rates of vocalizations change with excitatory state and how specific vocalizations vary independently of state and depend upon content. In this study, whistles (0.32 to 22 kHz) of two captive bottlenose dolphins (*Tursiops truncatus*) were recorded underwater before, during, and after six swim sessions in which the behavior of the dolphins and four human swimmers was controlled by a trainer on the dock. Sonograms of 205 min of recorded whistles were analyzed. Variation of whistle-type and rate was observed relative to the three contexts. The contours of distinguishable whistles ($n=591$) revealed 32 types. Ten types were observed during one context only; during the swim sessions. Almost half of the whistle types (15 to 32) were more likely to occur prior to swim sessions and the others were more likely during swim sessions. None were more frequent after swims. Rate of whistles was significantly higher than expected during the swim sessions ($z=7.21$) and lower than expected after the swim sessions ($z=-0.03$).

Session 4pPA

Physical Acoustics: Thermoacoustics and Bubbles

Sameer I. Madanshetty, Chair
Department of Mechanical and Aerospace Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Contributed Papers

2:30

4pPA1. Measurements with wire mesh stacks in thermoacoustic prime movers. Thomas J. Hofler and Mark S. Reed (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943)

Measurements with various wire mesh "stacks" in two different thermoacoustic prime movers are presented. Stirling engine regenerators are commonly constructed by stacking disks cut from wire mesh (i.e., wire cloth or wire screen) in a tube. In addition to simplicity, this has two advantages for prime movers. First, the wire is relatively impervious to moderately high temperatures and second, the effective thermal conductivity of the structure is one to two orders of magnitude lower than a comparable metal "parallel plate" structure. Since no linear theoretical thermoacoustic models exist for these wire mesh stacks, this approach is simply to measure the performance of several different mesh stacks. Initial results indicate reasonably good onset temperature and amplitude performance. [Work supported by Office of Naval Research.]

2:45

4pPA2. Measurements of transient effects in thermoacoustics. Albert B. Jones, Jr. (Dept. of Psych., Northeastern Univ., 1600 Massachusetts Ave., Boston, MA 02115), Anthony A. Atchley, David D. Hebert, Hsiao-Tseng Lin, Ching-Kai Meng, and Arthur R. Salindong (Naval Postgrad. School, Monterey, CA 93943)

Most theories of thermoacoustic phenomena deal with steady-state quantities. However, recent work by Prosperetti and co-workers is capable of treating the transient part as well. Motivated by these predictions, measurements are presented of the evolution of the temperature distribution along a stack plate exposed to a large amplitude standing wave and the evolution of the pressure waveform within a prime mover. Qualitative measurements of the former and both qualitative and quantitative measurements of the latter have been reported by various authors previously. The purpose of the present series of measurements is to provide a more complete set of data to test theoretical predictions. [Work supported by the Office of Naval Research.]

3:00

4pPA3. The heat transfer coefficient between stack and heat exchanger as a function of thermal penetration depth. James R. Brewster, Richard Raspet, and Henry E. Bass (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

The heat transfer coefficient between two elements of a thermoacoustic device is defined as being the ratio between the rate of heat flow between them and the temperature difference occurring at the interface. Previous experiments have demonstrated that this quantity is proportional to the amplitude of acoustic oscillation. For stack and heat exchanger duct widths that are comparable to the thermal penetration depth, the heat transfer coefficient was found to be equal to that predicted by a theory based on the elements of the engine being perfect heat exchangers. Results of experimental measurements of the variation of heat transfer coefficient as a function of frequency will be presented. Increasing the frequency reduces the thermal penetration depth allowing the effects of partial thermal contact within the elements to be evaluated. These results will be compared to attempts to build theoretical models of the heat exchange process.

3:15

4pPA4. Optimization of the performance of thermoacoustic refrigerators applying the short stack boundary layer approximation. M. Wetzel and C. Herman (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218-2686)

A systematic approach to optimize the performance of thermoacoustic refrigerators by using the COP (coefficient of performance) is presented. For this purpose four main modules of the thermoacoustic refrigerator were identified: (i) the thermoacoustic core, (ii) the resonance tube, (iii) the acoustic driver, and (iv) the heat exchangers. The main objective of the analysis was the optimization of the thermoacoustic core, which consists of the stack region of a thermoacoustic refrigerator, by applying the short stack boundary layer approximation. For the enthalpy and work flux equations of the short stack boundary layer approximation, which are the base for the calculations of the thermoacoustic core's COP, 20 independent design parameters were identified, and summarized in a multidimensional parameter space. Introducing a normalized multidimensional parameter space the number of design parameters was reduced to ten. Setting limits on the normalized design parameters allowed us to identify the three most significant design parameters: (i) normalized stack length, (ii) normalized stack center position, and (iii) normalized temperature difference. Theoretical calculations of the thermoacoustic core's COP indicate competitive values with commercially available refrigeration techniques. [Work supported by the Office of Naval Research; Martin Wetzel is also supported by a scholarship from DAAD.]

3:30–3:45 Break

3:45

4pPA5. Understanding the periodic driving pressure in the Rayleigh–Plesset equation. William C. Moss (Lawrence Livermore Natl. Lab., L-200, 7000 East Ave., P.O. Box 808, Livermore, CA 94550)

The radial motion of a single bubble in a periodically driven liquid is simulated by solving the Rayleigh–Plesset equation and the fully compressible hydrodynamic equations. The hydrodynamic equations require a much smaller far-field periodic driving pressure than the Rayleigh-Plesset equations to produce the same maximum bubble radius. The discrepancy is resolved by constructing analytic traveling and standing wave solutions that show the relationship between the far-field periodic driving pressure and the pressure near the bubble, which is actually responsible for the radial motion. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

4:00

4pPA6. An analytical and numerical study of nonlinear bubble oscillations in viscoelastic fluids. John S. Allen (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA 98195) and Ronald A. Roy (Univ. of Washington, Seattle, WA 98195)

The acoustically forced oscillations of gas bubbles in Newtonian fluids have been studied extensively; however, these oscillations have not been as thoroughly investigated in non-Newtonian fluids. An understanding of the non-Newtonian effects of viscoelasticity on this phenomena is particularly important to medical acoustic applications involving oscillatory bubble behavior in biological tissues and fluids. The stress tensor formalisms for non-Newtonian fluids used in previous works are reviewed and scrutinized. Inconsistencies in the literature and subsequent questions about the trace of the stress tensor are highlighted. As an initial step, a weakly nonlinear perturbation method is used to examine forced single bubble oscillations in a viscoelastic fluid. A novel analytical formulation is employed to make the perturbation approach (method of multiple scales) to the problem more tractable. Furthermore, this new formulation also allows for complementary numerical solutions. The physical limitations and relevance of this work with respect to medical applications are also discussed. [Work support by NIH through Grant No. RO1 CA39374.]

4:15

4pPA7. Intensified cavitation produced with pressure release and rigid ellipsoidal reflectors. Michael R. Bailey and David T. Blackstock (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029 and Mech. Eng. Dept., Univ. of Texas, Austin, TX 78712-1063)

An underwater bubble is well known to grow in response to a strong negative acoustic pulse and then collapse because of inertial forces. Here it is shown that adding an auxiliary positive pulse after collapse begins intensifies the collapse. The negative-then-positive pulse sequence is produced by two ellipsoidal reflectors, each with an electrical spark at its near focus f_1 and beamed so that they share a common second focus f_2. The negative pulse is produced by a polyurethane (pressure release) ellipsoid, the positive by a brass (rigid) ellipsoid. A timing circuit controls the delay between the pulses. Cavitation is recorded by pitting (caused by bubble collapse) of an aluminum foil membrane, which is centered at f_2 and lies coplanar with the two crossed beams. When the brass reflector is fired alone, a narrow path of ~ 1-mm diameter pits appears. Firing the polyurethane reflector alone yields more widespread, ~ 0.1-mm diameter pits. When both are fired, a pitted X pattern shows the position of the two beams. If the delay between the two firings is 2–6 μs, the intersection of and the centerline between the paths erupts with deep pits. [Work supported by ONR.]

4:30

4pPA8. Acoustic radiation from the monopole resonance of a bubble excited in a dielectrophoretic levitator by oscillating electric fields. Philip L. Marston, Nathaniel K. Hicks, and David B. Thiessen (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

Dielectrophoretic levitation of bubbles in an insulating liquid appears to be a novel way of studying acoustic radiation from transient bubble volume pulsations. Single bubbles were trapped in a spatially varying dc electric field designed to give stable levitation. A few cycles of a weaker oscillating field were superposed on the levitation field. When the modulation frequency was adjusted for the electric field to excite the monopole resonance, a transient acoustic signature was detected by an electrically shielded hydrophone in the surrounding oil bath. One mechanism for monopole excitation is that the energy of the bubble is increased by the electric field giving a positive electric pressure contribution proportional to the square of the field [P. L. Marston and R. E. Apfel, Phys. Lett. **60**A, 225–226 (1977)]. The same result follows from a surface average of the Maxwell stress using the Minkowski form of the stress tensor. Additional monopole coupling mechanisms may include a Helmholtz dielectric stress contribution (mitigated by the electrostrictive liquid response) and vertical oscillations of the bubble's centroid in the hydrostatic pressure field. [Work supported by the Office of Naval Research.]

4:45

4pPA9. The shape stability and interaction of millimeter-size gas bubbles trapped in an ultrasonic standing wave. Eugene H. Trinh (JPL/Caltech, MS 183-401, 4800 Oak Grove Dr., Pasadena, CA 91109)

A three-dimensional standing wave in a resonant liquid-filled cavity allows the trapping of multiple millimeter-size air bubbles. The coupling of the radiation pressure stresses to the bubble interfaces induces a net downward force counteracting gravity, a steady-state shape deformation, and it also drives natural shape oscillations of large amplitude under certain circumstances. The parameter space for the shape stability of trapped bubbles in ultrasonic fields having a frequency between 45 and 65 kHz has been experimentally determined in pure distilled water as well as in aqueous solutions containing surfactants. The variable parameters are the sound pressure, the sound wavelength, and the bubble equilibrium radius. The effects of the sound field intensity on the interaction of contacting and noncontacting neighboring bubbles are also experimentally investigated. In particular, a preliminary study of the impact of the sound intensity on the coalescence efficiency of contacting bubbles is conducted in pure water and in aqueous surfactant solutions. [Work funded by NASA.]

5:00

4pPA10. Collinear ultrasonic four-wave mixing in suspensions of particles and bubbles. Christopher S. Kwiatkowski and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

The interaction of sound with sound in a suspension can be mediated by the suspension's response to acoustic radiation pressure [H.J. Simpson and P.L. Marston, J. Acoust. Soc. Am. **98**, 1731–1741 (1995)]. Four wave mixing mediated by a suspension with collinear pump and probe waves is investigated. Radiation pressure from counter propagating pump beams induces a periodic grating in the number density of hollow glass microspheres suspended in a water/sugar mixture designed to make the particles neutrally buoyant. The Bragg scattering amplitude was measured using a single PVDF transducer as both the probe source and receiver that is parallel with the standing pump wave nodal planes. Reflection coefficients are calculated by comparing the reflected and transmitted waves. By increasing the particle concentration, total Bragg reflectivities from the established grating reaching 20% have been seen, which is still in the region of agreement between the Born approximation and transfer matrix theory. The dynamics of both the grating formation and dissolution have been studied, leading to the conclusion that grating dissolution is not dominated by diffusion alone. Bragg scattering by gratings of gas bubbles caused by cavitation was detected. [Work supported by the Office of Naval Research.]

Session 4pPP

Psychological and Physiological Acoustics: Auditory Capabilities and Proclivities of Normal and Impaired Listeners (Poster Session)

Sid P. Bacon, Chair

Department of Speech and Hearing Science, Arizona State University, P.O. Box 871908, Tempe, Arizona 85287-1908

Contributed Papers

All posters will be on display from 1:00 to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 to 5:00 p.m.

4pPP1. Acoustic reflex decay for modulated signals. Raymond D. Cook, Michael O. Ferguson, Joseph W. Hall III, John H. Grose, and Harold C. Pillsbury (Div. of Otolaryngol., Head and Neck Surgery, Univ. of North Carolina, 610 Burnett-Womack, CB# 7070 Chapel Hill, NC 27599)

Temporal decay of the acoustic reflex provides the basis for an objective audiological test that differentiates cochlear from retrocochlear pathologies. The classic sign of a neural lesion is a rapid decay of the reflex under conditions of pure-tone stimulation for frequencies 1000 Hz and below. This restriction to lower frequencies is due to the fact that even normal ears show decay for higher-frequency signals. At present, it is unclear whether the acoustic reflex decay (ARD) seen in normal ears is related to frequency-specific channels or whether the critical variable is the timing information coded within the channels. This study examined ARD in subjects with normal hearing and middle ear function. The degree of ARD was measured for both modulated and unmodulated carrier frequencies of 500, 1000, 2000, and 4000 Hz with modulation rates of 50 – 400 Hz. The dependent variable was the half-life of the decaying reflex (ARD 50%) over a 12-s stimulation interval. Significant ARD was present for high-frequency unmodulated carriers, but not for low-frequency carriers. For all listeners, ARD was diminished for all modulated stimuli, suggesting that resistance to ARD is mediated by the temporal characteristics of the stimulus. [Work supported by the American Hearing Research Foundation.]

4pPP2. Is higher acoustic energy at faster rates necessary for the click-rate-induced improvement in acoustic reflex thresholds? Vishakha W. Rawool (Commun. Disord. & Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

Ipsilateral click thresholds can improve by approximately 21 dB, with an increase in the click repetition rates from 50/s to 300/s [Rawool, Scand. Audiol. **24**, 199–205 (1995)]. When click-trains are presented over the same duration (e.g., 1.5 s), the number of clicks presented at the lower rates is less than the number of clicks presented at the higher repetition rates. Thus there is more total energy for the higher repetition rates. Therefore, it has been proposed that an important factor contributing to the threshold advantage for higher repetition rates could be the number of clicks. This investigation was designed to evaluate acoustic reflex thresholds at various repetition rates using a constant number of clicks (constant acoustic energy). Ipsilateral acoustic reflex thresholds were obtained from 19 normal female left ears by presenting 300 clicks at the repetition rates of 50, 100, 150, 200, and 300/s. The frequency of the probe tone was 226 Hz (85 dB SPL). The results showed a significant improvement (mean = 17 dB) in acoustic reflex thresholds with the increase in repetition rates. Thus the improvement in acoustic reflex thresholds with the increase in the repetition rate can occur as a function of the rate itself, even if a constant number of clicks are presented at each rate.

4pPP3. Effect of probe frequency on the click-rate-induced facilitation of the acoustic reflex thresholds. Vishakha W. Rawool (Commun. Disord. & Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

Ipsilateral click thresholds obtained with a 226-Hz probe tone can improve by approximately 21 dB, with an increase in the click repetition rates from 50/s to 300/s [Rawool, Scand. Audiol. **24**, 199–205 (1995)]. This investigation was designed to evaluate the effect of higher probe frequencies on the click-evoked acoustic reflex thresholds and on the rate-induced improvement in the acoustic reflex thresholds. Twelve female and six male subjects with normal hearing participated in the study. Ipsilateral acoustic reflex thresholds were obtained from the left ear of each subject by presenting clicks at the repetition rates of 60, 120, 180, 240, and 300 clicks/s. The results were obtained for three probe tone frequencies: 226, 678, and 1000 Hz. All the subjects showed a decrease in admittance for the 226-Hz probe and an increase in the admittance for the 1000-Hz probe. For the 678-Hz probe, some subjects showed an increase in admittance and others showed a decrease in admittance. The acoustic reflex thresholds were significantly elevated for the 678-Hz probe tone. The rate-induced improvement in acoustic reflex thresholds for each subject was calculated as the difference between the minimum and maximum acoustic reflex thresholds. This improvement (13 to 17 dB) was not significantly different for the three probe tones.

4pPP4. Effect of hearing impairment on the click-rate induced facilitation of the acoustic reflex thresholds in older individuals. Vishakha W. Rawool (Commun. Disord. & Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

Ipsilateral click thresholds can improve by approximately 21 dB, with in increase in the click repetition rates from 50/s to 300/s [Rawool, Scand. Audiol. **24** 199–205 (1995)] in young normal individuals. This improvement is significantly reduced in older individuals [Rawool, J Gerontol. Biol. Sci. (in press)]. The current investigation was designed to evaluate the effect of age-related hearing impairment on the rate-induced improvement in the acoustic reflex thresholds. Ipsilateral acoustic reflex thresholds were obtained from 60 older ears in the age range of 50 to 74 years. The subjects were divided into four groups: (I) Normal. Hearing within normal limits (20 dB). (II) Mid high-frequency impairment. Hearing within normal limits up to 2 kHz, and an average impairment of 17.5 to 25 dB at 3, 4, 6, and 8 kHz. (III) Moderate high-frequency impairment. Hearing within normal limits up to 2 kHz and an average impairment of 35 to 61 dB at 3, 4, 6, and 8 kHz. (IV) Flat hearing impairment. Hearing impairment across the test frequencies. Ipsilateral acoustic reflex thresholds were obtained from the left and/or right ear by presenting clicks at the repetition rates of 50, 100, 150, 200, and 300 clicks/s. No significant differences were apparent in the rate-induced improvement (11 to 14 dB) in the acoustic reflex thresholds across the four groups.

4pPP5. Middle ear of a lion: Comparison of structure and function to domestic cat. Gregory T. Huang, John J. Rosowski, Deborah T. Flandermeyer, and William T. Peake (Dept. of Elec. Eng. and Comp. Sci. and Res. Lab. of Elec., MIT, Cambridge, MA 02139 and Eaton-Peabody Lab., Mass. Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114)

Acoustic and anatomical measurements have been made in a deceased lion (*Panthera leo*) and compared to measurements in domestic cat (*Felis catus*). The results are used to test a model in which the acoustic impedance at the tympanic membrane (TM) is the sum of (1) the impedance of the TM and ossicular chain and (2) the impedance of the middle-ear cavities. Impedance was measured at the TM in lion before and after manipulation of the cavity structures; sound pressure in the cavities was measured simultaneously. CT sections of the lion middle ear were used to reconstruct the tympanic ring, the tympanic cavity, the bullar cavity, and the foramen that connects these cavities. A quantitative comparison of these structures shows that ear dimensions in the lion are roughly two times those in cats. Acoustic measurements in the lion and cat are qualitatively similar, but the lion middle-ear impedance is generally smaller. A six-element analog circuit model captures the main features of both sets of measurements. The acoustic effects of the size differences are represented by changes in the circuit model element values. In its ear, the lion is just a big pussycat. [Work supported by NIDCD.]

4pPP6. Relationship between the frequency microstructure of the pitch-level effect and the microstructure of synchronous evoked otoacoustic emissions. Edward M. Burns (JG-15, Univ. of Washington, Seattle, WA 98195)

The relationship between the frequency microstructure of the behavioral threshold and the frequency microstructure of the strength of synchronous evoked otoacoustic emissions (SEOAEs) and delayed evoked otoacoustic emissions is well documented [e.g., Zwicker and Schloth, J. Acoust. Soc. Am. **75**, 1148–1154 (1984)]. In a previous report from this laboratory [Burns *et al.*, J. Acoust. Soc. Am. Suppl. 1 **80**, S93 (1988)], it has been shown that there is also a correlation between the microstructure of the behavioral threshold and the microstructure of the pitch-level effect. In this paper, the relationship between changes in the microstructure of SEOAEs as a function of the level of the evoking stimulus and pitch shifts as a function of level is examined. [Work supported by NIDCD and the Virginia Merrill Bloedel Hearing Research Center.]

4pPP7. On the relation of distortion product and transient evoked emission spectral fine structure. Stefan Uppenkamp, Manfred Mauermann, and Birger Kollmeier (AG Medizinische Physik, Fachbereich Physik, Univ. Oldenburg, D-26111 Oldenburg, Germany)

Both distortion product emissions and transiently evoked emissions typically exhibit an individual fine structure, which seems to be linked to the fine structure of threshold in quiet observable in many normal hearing subjects. However, the recording paradigm for both experiments is rather different. Narrow-band distortion product emissions are detectable during continuous stimulation with two sinusoids. They represent the nonlinear interaction of the two tones in a stationary condition. Transient evoked emissions on the other hand may be interpreted as a type of impulse response of a nonlinear system. The frequency specificity is comparatively low. The interrelation of both types of emissions is not completely understood until now. To test the hypothesis of a common origin, patterns of spectral fine structure of DPOAE and TEOAE results from normal hearing subjects are compared. In addition, the influence of suppressor tones is investigated in both experiments. Results will be discussed in connection to current ideas about wave-fixed and place-fixed generation of emissions along the cochlear partition.

4pPP8. New evoked cochlear responses: Double chirp-evoked distortion products and double click-evoked otoacoustic emissions. Douglas H. Keefe (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

A novel family of evoked OAE responses controls probe distortion using a stimulus presentation method well-suited to research and clinical applications. The family allows a complementary representation between DP measurements and double click-evoked OAE measurements. Each stimulus sequence includes three, equal-duration sub-sequences defined as follows: $s_1(t)$ is a single chirp or click, $s_2(t)=\epsilon s_1(t-\tau)$ is a copy of s_1 with relative amplitude ϵ and delay τ $(0-5$ ms), and their superposition $s_{12}(t)=s_1(t)+s_2(t)$. The pressure response to each sub-sequence is p_1, p_2, and p_{12}, respectively. The double chirp-evoked distortion product (2ChDP) and double click-evoked OAE (2CEOAE) are defined by $p_D=p_{12}-p_1-p_2$. This subtraction of click respones is similar to Kemp *et al.* (1986) in eliminating the linear response, but differs in that particular choices of $\{s_1(t),\tau,\epsilon\}$ improve measurements of 2CEOAEs at high frequencies by eliminating time-gating, and at low frequencies by reducing probe distortion, especially when two acoustic sources are used. With a 16 kHz sample rate, 2ChDP and 2CEOAE responses have been obtained from 350-7900 Hz. A 2CEOAE special case is $\tau=0$, resulting in a wide bandwidth, single click-evoked OAE measurement without time-gating. Methodological refinements in real-time artifact rejection and nonlinear coherence help reduce noise. [Work supported by NIH Grant No. P01 DC00520.]

4pPP9. The development of forward masking in human infants. Lynne A. Werner (Dept. Speech & Hearing Sciences, Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246)

It has been previously reported that 3-month-old human infants are more susceptible to forward masking than are adults of the same species. The purpose of the present study was to determine whether susceptibility to forward masking declines between 3 and 6 months of age. Six-month-old infants were trained to respond when they heard a train of alternating bursts of a broadband noise and a 1-kHz tone, but not when they heard the noise bursts alone. The tone burst had 16-ms rise and fall times and no steady-state duration. The interval between noise offset and tone onset, Δt, was either 10, 25, or 200 ms; Δt was fixed for each infant. The onset to onset interval between noise bursts was constant at 350 ms + Δt on all trials. The level of the noise was 65 dB SPL; its duration was 100 ms. The level of the tone was manipulated to define masked threshold. Six-month-olds demonstrated about 10 dB more forward masking than adults and about 10 dB less forward masking than 3-month-olds for short Δt, and about the same amount of masking as 3-month-olds for 200 ms Δt. [Work supported by NIH.]

4pPP10. Role of fundamental frequency differences in perception of simultaneous vowels by hearing-impaired listeners. Kathryn Hoberg Arehart, Catherine Arriaga King, and Kelly S. McLean (CDSS Dept., Univ. of Colorado, CB 409, Boulder, CO 80309)

This study investigated the ability of normal-hearing listeners and listeners with moderate to moderately severe sensorineural hearing loss to use differences in fundamental frequency (F_0) in the identification of

monotically presented simultaneous vowels. Two psychophysical procedures, masked vowel identification and double-vowel identification, were used to measure identification performance as a function of F_0 differences (0 through 12 semitones) between simultaneous vowels. The masked vowel identification task yielded thresholds representing signal-to-noise ratios at which listeners could just identify target vowels in the presence of a masking vowel. Performance in the double-vowel identification task was measured by the percentage of trials in which listeners correctly identified both vowels in a double vowel. As previously reported, normal-hearing listeners showed a sharp increase in the ability to separate simultaneous vowels as F_0 differences increased from 0 to 1 semitone. Hearing-impaired listeners were less effective in using F_0 differences in the perception of simultaneous vowels. Reasons for hearing-impaired listeners' reduced performance probably include differences in their ability to use spectral and temporal cues across different frequency regions in their perception of complex-tone pitch. [Work supported by Deafness Research Foundation.]

4pPP11. Discriminability of two-component complex tones by normal-hearing and hearing-impaired listeners. Kathryn Hoberg Arehart and Peninah S. Fine (CDSS Dept., Univ. of Colorado, CB 409, Boulder, CO 80309)

The discriminability of complimentary two-tone complex tones (Voelcker pairs) was investigated in normal-hearing and hearing-impaired listeners using a 2AFC task. A two-component complex consists of two pure tones (f_1, f_2) which differ in frequency (Δf Hz) and intensity (ΔI dB). When $\Delta I = 0$ dB, the pitch of the complex corresponds to the center frequency (f_c) where $f_c = (f_1 + f_2)/2$. When $\Delta I > 0$ dB, the pitch shifts toward the more intense component. In one tone of a Voelcker pair, the intensity of $f_1(I_1)$ is greater than the intensity of $f_2(I_2)$ by ΔI; in its complement, $I_1 < I_2$ by ΔI. Discriminability of the pitches of Voelcker pairs was measured as a function of Δf (10 through 1000 Hz), for three f_cs (1000, 2000, 4000 Hz) and for two ΔIs (1, 3 dB). Normal-hearing listeners' results were similar to previous results [Feth *et al.*, J. Acoust. Soc. Am. **72**, 1403–1412 (1982)]. Thresholds for discrimination of Voelcker pairs by hearing-impaired listeners were abnormally large across all center frequencies. Results will be discussed in terms of predictions of EWAIF/IWAIF models [Anantharaman *et al.*, J. Acoust. Soc. Am. **94**, 723–729 (1993)] and hearing-impaired listeners' deficits in using temporal cues in pitch perception.

4pPP12. Acoustic properties of the chinchilla pinna and ear canal. William J. Murphy and Rickie R. Davis (Bioacoust. and Occup. Vib. Sect., Natl. Inst. for Occup. Safety and Health, MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

Measurements of the acoustic transfer function (ATF) of the pinnae of 4 chinchillas were compared with the auditory-evoked potential (AEP) hearing thresholds of 16 chinchillas measured in free field and with insert earphones. The ATF was measured in anesthetized chinchillas in a far-field condition in a semi-anechoic room. Probe microphone measurements were collected just outside the pinna and at the tympanic membrane. The ATF exhibited a broad resonance between 2 and 6 kHz with about a 10 dB gain. The chinchillas were monauralized and had a chronic electrode implanted in the left inferior colliculus. The animals were awake and restrained during the AEP testing. The AEP measurements were averaged from 512 presentations of a toneburst at 0.5, 1, 2, 4, and 8 kHz. AEPs collected with insert earphones were calibrated with a probe microphone positioned near the tympanic membrane. The differences in the AEP hearing thresholds measured in the different configurations exhibited a 10 dB resonance at 4 kHz. The agreement between the ATF and AEP-derived transfer function suggested that the threshold differences from the two testing configurations could be accounted for by the pinna and ear canal gain.

4pPP13. Age-related hearing loss, temporary threshold shift and permanent threshold shift in four strains of mice. Yea-Wen Shiau, Ernest M. Weiler, Laura Kretschmer, Angel Dell'aira Ball (ML# 379, Psychoacoustics Lab. Communication Sci. & Disorders, Univ. of Cincinnati, Cincinnati, OH 45221), and Mary Anne Baker (Indiana Univ. Southeast, New Albany, IN)

A total of 96 mice from two inbred and two hybrid strains were tested for TTS (temporary threshold shift) and PTS (permanent threshold shift) using electrophysiological procedures for auditory brain-stem responses. Testing continued for up to 3 months following 2 h of exposure to 110-dB broadband noise. The inbred and hybrid strains carrying genes for age-related hearing loss showed TTS almost equal to their PTS "thus anticipating their tendency to hearing loss." On the other hand, the inbred and hybrid strains not carrying those genes showed considerable recovery from their original TTS losses. Those with age-related hearing genes were very susceptible to noise-induced PTS. An additional effect was the presence of strong correlations between TTS and PTS for both inbred strains, but no correlation for the two hybrid strains. Noise exposure history plays a role in the noncorrelations between TTS and PTS typically found in humans. Could genes influence correlation between TTS and PTS in humans?

4pPP14. Hearing and development impairment in Down-syndrome children. M. Angela G. Feitosa and Rosana M. Tristao (Instituto de Psicologia, Univ. Brasilia, Brasilia, DF, 70910-900, Brazil)

A growing number of studies have shown that Down-syndrome persons have a high incidence of auditory deficit and slow language development. Down-syndrome children were investigated with respect to (a) hearing level, (b) language development, (c) global development, and (d) environmental interaction. Fifty-one children, 6-47 months old, were observed, 22 of whom had Down syndrome. Controls included 13 children without diagnosis of mental impairment, and 16 children with mental impairment due to etiologies other than Down syndrome. None of the subjects had a previous diagnosis of hearing loss. Relations among levels of hearing, global and language development, home stimulation, and risk factors for hearing loss were analyzed. The Down-syndrome group showed significantly elevated auditory thresholds and a larger number of pathological findings in the external and middle ears, as compared to other groups. It also showed lower levels in language development, as compared to global development; and shortened latencies of waves I and V in BERA. [Work supported by CNPq and CORDE.]

4pPP15. The effect of head protectors on warning sound perception in noisy workplaces. Martin Fortin, Raymond Hetu, Hung Tran Quoc, and Stephane Denis (Groupe d'acoustique de l'universite de Montreal, C.P. 6128, Succ. Centre-ville, Montreal, PQ H3C 3J7, Canada)

The acoustic response of head protectors was assessed by means of a mecano-acoustic head simulator. The following protective devices were tested: welders' mask, fire-fighter helmet, aluminized hood, leather hood, sand-blasting hood, dust protection hood, and emergency respirator. The tests were conducted in an hemi-anechoic chamber with a wideband noise presented at 100 dB SPL. The effect of 36 horizontal combined with 13 vertical angles of incidence was assessed. Generally speaking, head protectors are responsible for insertion gains of 5 to 10 dB in the low frequencies and for insertion losses of up to 30 dB in the high frequencies. The acoustic response of head protectors considerably varied with the angle of incidence. With the welder's mask, when the source is coming from the rear, a systematic gain is observed at almost all frequencies; the mask acts as a low-pass filter in the frontal quadrant. These findings hold implications with respect to health and safety issues. In term of risk of hearing damage, the wearing of a head protector may significantly increase the noise exposure level of a worker. In terms of safety, such devices may interfere with signal detection in noise and sound source localization. [Work supported by IRSST.]

4pPP16. Physiological correlates of the time-intensity trade in auditory late responses. Joelle Redner and Ann Clock Eddins (Dept. of Speech and Hearing Sciences, Aud. Phys. Lab., Indiana Univ., Bloomington, IN 47405)

Numerous psychophysical studies have demonstrated that behavioral detection thresholds decrease with increasing stimulus duration, often referred to as the time-intensity trade. Few studies, however, have evaluated the physiological correlates of this trading relationship. The purpose of the present study was to examine the effect of stimulus duration on evoked potential thresholds known as the auditory late response (ALR). Behavioral and ALR thresholds were estimated for 1000 and 4000 Hz tone bursts at 8, 16, 32, 64, and 128 ms in five normal-hearing, young adults. Evoked responses were recorded in 2-dB steps from 20 dB to −4 dB SL *re:* behavioral threshold. ALR threshold was estimated as the lowest level at which the N1-P2 complex was visually detected by two independent judges. Preliminary results show that both behavioral and ALR thresholds decreased as a function of increasing duration. For the 1000-Hz stimulus, both ALR and behavioral thresholds decreased by approximately 23 dB HL over the range of durations tested. Average ALR and behavioral thresholds for the 4000-Hz stimulus decreased by about 19 dB HL. These results show that the time-intensity trade can be demonstrated physiologically using the auditory late response.

4pPP17. Detection of frequency-modulated signals by cochlear-implant users. Ina Rea Bicknell and Lawrence L. Feth (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

Many cochlear-implant users show postoperative improvement in the ability to hear speech but most rely on lip-reading to understand speech. To determine if this limitation on speech perception relates to impaired temporal resolution, and, thus, an inability to track rapid frequency changes which are characteristic of speech, temporal resolution thresholds (TRT) of adult implant users were determined by measuring their ability to discriminate between a glide signal, a sinusoid linearly modulated in frequency and a step signal, a sinusoid traversing the same frequency range but in two to five discrete steps. Performance of implanted subjects was extremely variable, but, generally, their TRTs were higher than those of age- and sex-matched normal-hearing controls. At certain frequencies and sweep extents, no implanted subjects were able to distinguish between step and glide signals of duration shorter than 300 ms. When the signal step number was no greater than two, one subject, using a Spectra speech processor, had TRTs near the 10 ms upper limit of the control TRTs. Extended frequency sweeps of 1350–1500 Hz with center frequencies of 3300–3400 Hz resulted in nearly normal TRTs for two of the implant users. [Work supported by the Dayton, OH VAMC and a grant from AFOSR.]

4pPP18. Auditory brain-stem responses in adults with chronic conductive hearing loss. Michael O. Ferguson, Raymond D. Cook, Joseph W. Hall III, John H. Grose, and Harold C. Pillsbury (Div. of Otolaryngol., Head and Neck Surgery, Univ. of North Carolina, 610 Burnett-Womack, CB# 7070, Chapel Hill, NC 27599)

The ABR is a brief latency, electrophysiologic response that reflects synchronous neuronal activity from the auditory-nerve and brain-stem structures. Investigations have shown that children with early conductive impairment exhibit abnormalities in their ABR, suggesting that the nervous system has an element of plasticity during critical periods of development. However, no similar research has been undertaken on adults to determine whether ABR abnormalities associated with chronic conductive impairment also exist in this population. This study addresses that question by investigating the ABR in a group of adults with chronic conductive loss and a control group of normal-hearing adults. The goal was to determine whether chronic conductive hearing loss in adults would result in similar abnormal ABR measurements as seen in the juvenile population. Subjects were evaluated by comparing ABR interpeak latencies between waves I and III and waves I and V. In addition, interaural latency differences were assessed in cases of unilateral disease. Preliminary results suggest that ABR abnormalities associated with conductive impairment are present. Specifically, in the cases of unilateral disease,there is a significant difference between ears in the waves I–V interpeak latency which is independent of audiometric asymmetry. These data may indicate that an element of plasticity exists in the mature auditory system. [Work supported by NIH NIDCD.]

4pPP19. Estimates of spectral weights and internal noise in the discrimination of spectral variance. Robert A. Lutfi and Eunmi Oh (Dept. of Communicative Disorders, Univ. of Wisconsin, Madison, WI 53706)

An experiment was conducted to measure the relative contribution of spectral weights and internal noise in the discrimination of spectral variance. The stimuli were simultaneous tone complexes comprised of the six octave frequencies from 250 to 8000 Hz. On each presentation the levels of components in dB were drawn independently and at random from one of two normal distributions having identical means but different variances ($\sigma_N = 1$ dB, $\sigma_S = 2$–10 dB). In the standard 2IFC procedure, the listener's task was to choose the complex having the greater variance in component level. The shape of the psychometric function for all five listeners was markedly different from that of an observer limited only by additive internal noise. It was consistent with an observer that gives weight to only one or two components in the complex. This result, however, was inconsistent with the weighting functions computed from the trial-by-trial data from these listeners. Both measures can be reconciled if it is assumed that listener weights vary from trial to trial, or equivalently that observations are corrupted by multiplicative rather than additive internal noise. [Work supported by NIDCD RO1 DC01262-04.]

4pPP20. Intrinsic envelope fluctuations and modulation-detection thresholds for narrow-band noise carriers. Torsten Dau, Jesko Verhey (Graduiertenkolleg Psychoakustik, Fachbereich Physik, Univ. Oldenburg, D-26111 Oldenburg, Germany), and Armin Kohlrausch (Inst. Percept. Res. (IPO), Eindhoven, The Netherlands)

This contribution summarizes analytical properties of envelope modulation spectra of bandpass noise signals, which are relevant for the dependence of amplitude-modulation thresholds on modulation rate and noise carrier bandwidth. Lawson and Uhlenbeck [*Threshold Signals* (McGraw-Hill, New York)] showed in 1950 that the modulation spectrum, i.e., the power spectrum of the linear envelope of a bandpass noise, has an approximately triangular continuous spectrum besides the dc peak. For a constant overall level of a noise band, the total power of intrinsic noise fluctuations, i.e., the total area under the triangle, remains constant. The modulation spectrum, however, becomes broader with increasing noise bandwidth. For low modulation rates, this leads to a reduction in intrinsic modulation power. For higher modulation rates, the intrinsic modulation power increases. Simple analytical calculations are presented which compute the integrated modulation power of the noise within the transfer function of a hypothetical modulation filter. For a set of carrier bandwidths between 1 and 6000 Hz and for signal modulation of 5, 25, and 100 Hz the calculated values of overall modulation power are compared with experimentally obtained modulation detection thresholds, and with results from simulations of an auditory model.

4pPP21. Intensity weighted average of instantaneous frequency computations using time-frequency representations. Jayanth N. Anantharaman, Ashok K. Krishnamurthy (Dept. of Elec. Eng., Ohio State Univ., Columbus, OH 43210), and Lawrence L. Feth (Ohio State Univ., Columbus, OH 43210)

The intensity weighted average of instantaneous frequency (IWAIF) has been used to model listener performance in complex signal discrimination [Anantharaman *et al.*, J. Acoust. Soc. Am. **94**, 723–729 (1993)]. Short-term [Krishnamurthy and Feth, J. Acoust. Soc. Am. **93**, 2387(A)

(1993)] and multichannel [Mokheimer *et al.*, IEEE Proc. First Intl. Conf. on Electronics Circuits and Systems, Cairo, Egypt (1994)] implementations of the model have been proposed. Time-frequency representations (TFRs) offer a convenient way to process signals over the time-frequency plane. TFRs can be used to compute the IWAIF. TFRs can also be used to compute the short-term IWAIF and the multi-channel IWAIF. These will be presented along with preliminary developments of both the short-term IWAIF and the multichannel IWAIF. The necessary properties of the TFRs to be suitable for use in these models will be discussed. [Work supported by a grant from AFOSR.]

4pPP22. A quantitative prediction of modulation masking with an optimal-detector model. Torsten Dau, Birger Kollmeier (Graduiertenkolleg Psychoakustik, Fachbereich Physik, Univ. Oldenburg, D-26111 Oldenburg, Germany), and Armin Kohlrausch (Inst. Perc. Res. (IPO), Eindhoven, The Netherlands)

A multichannel model is discussed which describes effects of spectral and temporal integration in amplitude-modulation detection and masking. Envelope fluctuations within each auditory channel are analyzed with a modulation filterbank. The parameters of the filterbank are the same for all auditory filters and were adjusted to allow the model to account for modulation detection and modulation masking data with narrow-band carriers at a high center frequency. In the detection stage, the outputs of all modulation filters from all excited peripheral channels are combined linearly with optimal weights. To integrate information across time, a "multiple-look" strategy is implemented within the detection stage. Model predictions are compared with literature data from Houtgast [J. Acoust. Soc. Am. **85**, 1676–1680 (1989)]. The following three conditions were reproduced with a deviation of less than 3 dB between experiment and simulation. (1) Masking of test modulations in the range 2 to 64 Hz by a narrow-band masker modulation at 4, 8, or 16 Hz. (2) Modulation masking as a function of the masker-modulation level. (3) Modulation masking as a function of the masker-modulation bandwidth. The results from the simulations further support the hypothesis that amplitude fluctuations are processed by modulation-frequency-selective channels.

4pPP23. Comodulation masking release as a function of masking noise-band temporal envelope similarity in normal hearing and cochlear impaired listeners. Lee Mendoza, Mari L. Schulz, and Richard A. Roberts (Dept. of Speech Pathol. and Audiol., Univ. of South Alabama, UCOM 2000, Mobile, AL 36688)

Thresholds for a pure-tone signal (1000 Hz) were obtained from both normal hearing and hearing-impaired listeners in a variety of masker conditions. Masking stimuli consisted of five Gaussian noise bands, each 20 Hz wide, and centered on 500, 750, 1000, 1250, and 1500 Hz. Two such base stimuli were created. In the first stimulus set, all noise bands had the same temporal envelope (comodulated). In the second set, each noise band was generated independently of the other bands, and thus each had independent temporal envelopes. Additional masking stimuli were generated by combining the comodulated and independent bands at specific comodulated/independent intensity ratios (CIR = 25, 20, 15, 10, and 5 dB), with overall level of the combined noise bands held constant. The result of increasing CIR was a progressive increase in the similarity of temporal envelopes of the noise bands masking the signal. Compared to threshold for the pure-tone signal in independent bands of noise, threshold steadily decreased as the CIR increased for normal hearing listeners. Cochlear-impaired subjects also showed decreased threshold with increasing CIR; however, the improvement was seen to plateau at relatively low CIRs. [Work supported by DRF and USARC.]

4pPP24. Discrimination of harmonic- and log-spaced profiles and of static and dynamic profiles by good and poor profile listeners. Ward R. Drennan and Charles S. Watson (Dept. of Speech and Hear. Sci., Indiana Univ., Bloomington, IN 47405)

Most profile experiments have employed static profiles with logarithmic spacing. Many naturally occurring sounds have harmonic spacing and vary dynamically in time. Watson and Drennan [J. Acoust. Soc. Am. **97**, 3272(A) (1995)] examined profile discrimination using both static and frequency-glide profiles with harmonic and logarithmic component spacing. Subjects detected an intensity increment in the middle component of 11-component, 400-ms profiles with a frequency range of 200–2200 Hz. Differences among the seven subjects were considerably larger than differences among the types of profiles, confirming earlier observations of a large range of abilities to discriminate profiles. Another experiment was therefore conducted to estimate the distribution of profile discrimination abilities for normal-hearing listeners. Forty-six subjects were screened using static-log profiles. The distribution of thresholds was roughly normal with a range of -2 to -26 dB (signal level relative to component level) and an s.d. of 4.8 dB. No dichotomy in profile discrimination ability was found. Subjects were selected from each tail of the distribution and tested using the static-log, static-harmonic, dynamic-log, and dynamic-harmonic profiles. The order of presentation of conditions significantly affected the results; however, whatever the order, static profiles yielded lower thresholds than the frequency-glide profiles. [Work supported by NIH/NIDCD and AFOSR.]

4pPP25. Distinctiveness and serial position effects in tonal sequences: Combining DDT and PTD. Aimee M. Surprenant (Dept. Psychol. Sciences, Purdue Univ., W. Lafayette, IN 47907)

The proportion-of-the-total duration rule (PTD) [Kidd and Watson, J. Acoust. Soc. Am. **92**, 3109–3118 (1992)] states that the detectability of a change in a component of a tonal sequence can be predicted by the duration of the changed component relative to the sequence as a whole. A similar idea has been used in dimensional distinctiveness theory (DDT) [Neath, Mem. Cognit. **21**, 689-698 (1993)] to account for primacy, recency, and other serial position effects in memory. An item will be remembered if it is more distinct along some dimension relative to possible competitors. These experiments explore the relation between DDT and PTD by examining the effect of inter-stimulus interval (ISI) on the detection of a change in one tone of a tonal sequence for each serial position in the sequence. Results show that, with increasing ISI, performance on the first items increases whereas performance on the final items decreases, as predicted by DDT. In addition, the interaction of PTD and ISI was explored for each serial position. This research combines theories proposed in the psychophysical and memory areas and suggests that a comprehensive principle based on relative distinctiveness can account for both perceptual and memory effects.

4pPP26. Temporal integration as a function of masker bandwidth. Andrew J. Oxenham (Inst. for Percept. Res. (IPO), P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

Thresholds were measured for a 6-kHz sinusoid, temporally centered in a 500-ms masker which was either a bandpass Gaussian noise (20 dB SPL spectrum level) or a 6-kHz sinusoid (40 dB SPL). A notched noise centered on 6 kHz prevented the use of off-frequency cues. The signal, gated with 2-ms ramps, ranged in half-amplitude duration from 2 to 300 ms. The noise bandwidth was arithmetically centered on 6 kHz and was varied from 60 Hz to 12 kHz. For masker bandwidths below 300 Hz, the slope of integration for signal durations between 2 and 20 ms decreased with decreasing masker bandwidth. For the tonal masker, increasing signal duration from 2 to 20 ms had no effect on threshold. These results cannot be accounted for by lowpass-filter or temporal-window models of temporal integration or resolution. Instead, it is proposed that the auditory system performs a spectral analysis of the stimulus envelope, so that the rapid fluctuations (high modulation frequencies) introduced by gating the signal

can be used as a cue for brief signals in narrowband noise and tones. For broadband maskers, this cue is not available due to the masker's inherent rapid envelope fluctuations. [Work supported by the Wellcome Trust.]

4pPP27. Sinusoidal amplitude modulation thresholds as a function of carrier frequency and level. Ralf Fassel and Armin Kohlrausch (Inst. for Perception Res., P.O. Box 513, NL-5600 MB Eindhoven, The Netherlands)

Modulation detection thresholds for sinusoidal carriers were obtained for a wide range of modulation rates (10–1600 Hz) as a function of carrier frequency (1, 3, 5, 8, and 10 kHz) and carrier level (30, 45, 60, and 75 dB SPL). At low modulation rates, between 10 and about 100 Hz, thresholds were roughly constant. For higher modulation rates, thresholds were dependent on carrier frequency. For the lowest carrier frequency (1 kHz), the modulation sidebands were already spectrally resolved at a modulation rate of 100 Hz and thus thresholds decreased with increasing modulation rate. For the higher carrier frequencies, the sidebands were resolved only at higher modulation rates, due to the increasing auditory-filter bandwidth. In these cases, thresholds initially increased with increasing modulation rate with a slope of about 8 dB/oct. The threshold curves show a different shape than those obtained with broadband noise carriers. In general, thresholds decreased with increasing carrier level at any given carrier frequency up to at least 60 dB SPL. For one subject a saturation of thresholds at medium carrier levels was observed. At very low carrier levels of 20 dB SL, thresholds tended to increase with increasing modulation rate already below 100 Hz.

4pPP28. Across-channel processes in frequency modulation detection. Shigeto Furukawa and Brian C. J. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, UK)

This study examined whether the detection of frequency modulation (FM) on two carriers depends on the coherence of the FM across carriers. Psychometric functions were measured for detecting sinusoidal FM of carriers with frequencies 1100 and 2000 Hz. The modulators for the two carriers were either in phase (coherent) or in anti-phase (incoherent). The modulation rate was either 2.5, 5, or 10 Hz. One or more cycles of modulation were used. The modulation of each carrier was equally detectable, as determined in a preliminary experiment. A continuous pink noise background was used to mask the outputs of auditory filters tuned between the two carrier frequencies. Detectability was better for coherent FM than for incoherent FM. The effect of FM coherence was greatest at the lowest modulation rate, possibly indicating that phase locking plays a role [B. C. J. Moore and A. Sek, J. Acoust. Soc. Am. **97**, 2468–2478 (1995)]. The detectability of the coherent FM was well above the value predicted on the assumption that information from the two carrier frequencies was processed independently and combined optimally. These results imply the existence of one or more mechanisms sensitive to FM coherence. [Work supported by the MRC(UK), the British Council, and an ORS award.]

4pPP29. Within and across channel processes contributing to comodulation detection differences. Stephen J. Borrill and Brian C. J. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, UK)

The threshold for detection of a narrow-band noise signal was determined in the presence of a masker consisting of two synchronously gated 20-Hz-wide bands of noise with a spectrum level of 65 dB which were comodulated. The maskers were spaced by ΔF Hz above and below the signal frequency (1500 Hz). A low-pass noise was used to mask any combination products. Three conditions were tested: signal and masker envelopes correlated (C), signal and masker envelopes uncorrelated (U), or sinusoidal signal of the same overall level (S). ΔF ranged from 200 to 1400 Hz. Masked thresholds for the U and S conditions were essentially identical for all ΔF. Thresholds were higher in the C condition, thus showing a comodulation detection difference (CDD). The CDD was greatest for ΔF between 400 and 600 Hz. A second set of conditions was used in which either one, both, or none of the masking bands was comodulated with the signal. This was done for three different masker spectrum levels at a ΔF of 600 Hz. The results suggest that the CDD seen in the first experiment was mainly due to the upward spread of masking from the lower band and not to an across-channel grouping effect.

4pPP30. Amplitude-modulation depth discrimination of a sinusoidal carrier. Jungmee Lee and Sid P. Bacon (Psychoacoustics Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-1908)

Discrimination of the change in depth of sinusoidal amplitude modulation was investigated for a 4000-Hz carrier. The just noticeable change in the modulation depth (Δm) was measured as a function of (1) standard modulation depth ($m = 0.1, 0.18$, or 0.3), (2) modulation rate ($f_m = 10$, 20, 40, or 80 Hz), and (3) stimulus duration (T=25, 50, 100, 200, 400, or 800 ms). For modulation rates less than 80 Hz, threshold (Δm) was higher at a standard depth of 0.3 than at the other standard depths. When $f_m = 80$ Hz, the threshold was almost the same across different standard modulation depths. For all standard depths and modulation rates, the threshold decreased by more than a factor of 2 as stimulus duration increased to a certain T (critical duration). For durations longer than the critical duration, the threshold decreased only slightly or remained constant. The critical duration corresponded to about four cycles of modulation. Psychometric functions were measured for different stimulus durations. The data were evaluated in terms of a multiple-looks model. [Work supported by NIDCD.]

4pPP31. Detection of tones in modulated noise: Effects of masker level and masker depth. Sid P. Bacon, Jungmee Lee, Daniel N. Peterson, and Dawne Rainey (Psychoacoustics Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-1908)

It is possible to estimate temporal resolution at discrete spectral locations by subtracting the masked threshold produced by a modulated masker from that produced by an unmodulated masker (the difference is referred to as the modulated-unmodulated difference, or MUD). This paradigm may be especially useful for measuring temporal resolution in subjects with hearing loss, provided that the MUD is independent of level. The purpose of the present study was to examine the MUD as a function of masker level at several signal frequencies. In the first experiment, the sinusoidally amplitude-modulated masker had a depth (m) of 1.0. The MUD increased by as much as 15 dB as the spectrum level of the masker increased from 0 to 40 dB SPL. In the second experiment, the modulated masker had a depth of 0.75 or 1.0. When the masker depth was 1.0, the MUD increased with increasing masker level, as in experiment one. When it was 0.75, however, the MUD — though reduced — was essentially independent of masker level. These results suggest that a masker depth of 0.75 may be used to compare temporal resolution between normal-hearing and hearing-impaired subjects without being complicated by effects of masker level. [Work supported by NIDCD.]

4pPP32. Effects of noise on the hearing system. King Chung (Dept. of Audiol., Northwestern Univ., 2299 N. Campus Dr., Evanston, IL 60201)

This paper provides an overview of effects of noise on the hearing system. Animal research has shown that excessive noise exposure results in decrease of blood flow in stria vascularis, alteration in permeability of ions in reticular lamina, swelling/rupture of hair cells, disarrangement or loss of stereocilia, broken tip links and side links between the stereocilia, swelling of afferent dentrides, rupture of the organ of Corti, decrease in tectorial membrane thickness, and increase in its compliance, etc. After the exposure, the stereocilia may become a fanlike structure or fuse together to form a giant stereocilia. In a severely damaged cochlea, the hair cells may

be replaced by some undifferentiable scar tissue. The pharyngeal processes of the Deiter's cells and other supporting cells enlarge to seal off the endolymph and perilymph ion boundaries. The neural fibers and ganglion cells may gradually degenerate and tonotopic reorganization in dorsal cochlear nucleus follows. Along with these anatomical damages, physiological changes include disruption of the cochlear active process, changes in cochlear potentials, broadening of the turning curve of basilar membrane and spiral ganglion neurons, and hearing loss are also observed. Remedial procedures, e.g., regeneration of hair cells and "preexposure conditioning" are also discussed.

THURSDAY AFTERNOON, 16 MAY 1996 CELEBRATION A, 1:30 TO 5:10 P.M.

Session 4pSA

Structural Acoustics and Vibration: Memorial Session for Elfyn Richards

Joseph M. Cuschieri, Chair
Department of Ocean Engineering, Center for Acoustics and Vibration, Florida Atlantic University, Boca Raton, Florida 33431

Chair's Introduction—1:30

Invited Papers

1:35

4pSA1. The foundation of the Institute of Sound and Vibration Research. Peter O. A. L. Davies (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO17 1BJ, UK)

There are few people for whom a flourishing research institution provides a living memorial. That the ISVR was born at all was due to the characteristic and legendary "ten-year vision" of its founder, Professor Elfyn Richards. In the 1950's, as Professor of Aeronautics at Southampton, he perceived that noise and vibration were to become a "disease" of new technology, for which traditional approaches would be no cure. Thus he founded a new applied science, now known as engineering acoustics and vibration. This memorial paper describes the first two decades of both the foundation and early development of the ISVR. It concludes with a brief appraisal of its current activities.

2:00

4pSA2. Jet noise: The early history. Geoffrey M. Lilley (Dept. of Aeronautics and Astronautics, Univ. of Southampton, Southampton SO 17 1BJ, UK)

The invention of the jet engine and its development during World War II as the propulsive system for fighter aircraft led aircraft designers in the late 1940s to investigate its use as the power plant for civil air transports. The aim was improved speed, economics, and efficiency over that of slower propellor-driven aircraft. However two major problems existed, (a) its high fuel consumption, and (b) its noise at full take-off power. Innovative University jet noise research in the UK in the late 1940s and early 1950s quickly led to strong industry–university collaboration. This resulted in the development of the corrugated or lobed nozzle for jet noise reduction and later to the introduction of the bypass engine achieving significant reductions in both specific fuel consumption and jet noise reduction. The experiments by R. Westley and G. M. Lilley at Cranfield and by Professor E.J. Richards and A. Powell at Southampton University, Professor Sir James Lighthill's theoretical studies at Manchester University, and the skill of F.B. Greatrex and his team at Rolls-Royce all contributed to this successful revolution in civil aircraft design. The paper discusses the rapid development of this early jet noise research and the corresponding complementary work in the USA.

2:25

4pSA3. E. J. Richards and jet noise research at Southampton 1951–56. Alan Powell (Dept. of Mech. Eng., Univ. of Houston, TX 77204-4792)

Elfyn Richards became the first Professor of Aeronautical Engineering at University College, Southampton, in September 1950 and he immediately began planning for jet noise research. The author joined him in January 1951, remaining until August 1956. Some events of that period are reviewed. With Richards' boundless energy and infectious enthusiasm, his indispensible help with more than a touch of impatience—kindly toward the author, less so for administrative barriers—progress was remarkably rapid, though the experimental apparati were in most respects very modest by later standards. The eighth power law for turbulent jet noise power (and the awkward lack of proportionality of peak frequency with velocity), the screech of choked jets and supersonic edgetones were all discovered in the first year. Feedback theories for the latter two, pulse jet noise, and other investigations such as shock wave interaction and means of reducing screech were soon to follow. Acoustically inducted fatigue (AIF) of aircraft structures caused by jet noise was predicted and soon after was found to occur in practice. Richards played some role in all of these topics, but his innumerable ideas and widening noise interests could not possibly all be addressed in that period.

2:50

4pSA4. Encounters I had with Professor Richards. G. Maidanik (David Taylor Model Basin, NSWC, Bethesda, MD 20084-5000)

I met Professor Richards for the first time in 1951, when I was considering Southampton University for undergraduate studies. He never remembered it, I have. The second time I met Professor Richards was in 1962 in Boston, and then again and again over the subsequent years. I was always impressed by his keen scientific sense and his friendliness, the mark of a great educator, which he certainly was.

3:15–3:30 Break

3:30

4pSA5. Recalling some contributions by Elfyn Richards. Stanley E. Dunn (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

Professor Elfyn Richards enjoyed a long and unusually notable career which ranged over engineering, education, and public policy development. The scope of his impact was international and, as the Honorary Fellow award by the Society proclaims, of particular importance in the United States. Toward the end of his career he made himself available to work with the Ocean Engineering Department at Florida Atlantic University. That interaction was to have a profound impact on the program and its subsequent success in a number of areas. This paper shares the highlights of Professor Richards' collaboration with the department and discusses the benefits which were so dependent on the unique qualities of his character.

3:55

4pSA6. Structural damping by granular materials. Arcanjo Lenzi (Lab. de Vibracões e Acústica, Dept. Eng. Mecânica, Univ. Federal de Santa Catarina, C.P. 476, CEP 88037-140, Florianopolis SC, Brazil)

Studies on a useful and rarely used damping technique, the filling of cavity components with granular materials, are presented. Experimental studies show regions of maximum damping caused by standing wave formation in the material. The first maximum occurs when the internal dimension of the cavity is equal to one quarter of the longitudinal wavelength. The tuning of optimum damping to any desirable frequency requires accurate knowledge of the wave speed. Experiments show that wave speeds decrease with amplitude when strains reach values higher than about 10^{-6}. This is caused by gross slip taking place at contacts which breaks the main structure of grain–grain contacts responsible for the propagation of elastic waves. Waves speed whose amplitude produce strain smaller than 10^{-6} were observed being independent of amplitude. Loss factors of granular materials also present identical variation with strain. At large strains ($>10^{-6}$) energy is mainly dissipated at grain contacts, by dry friction in partial and gross slip forms. For low strains, energy is dissipated by hysteresis inside the grains themselves. Applications of this technique to heavy structures are useful.

4:20

4pSA7. Experimental statistical energy analysis: A tool for the reduction of machinery noise. N. Lalor (ISVR, Univ. of Southampton, Highfield, Southampton SO17 1BJ, UK)

The use of statistical energy analysis (SEA) for the solution of vibroacoustic problems that occur with the operation of machinery, has only recently become practicable. This paper charts the developments that led from the early ideas of energy accountancy (EA) to the current SEA modeling techniques. The basic concept of EA, which was conceived to show the influence of the overall design parameters on the radiated noise, is explained and its limitations discussed. The paper then goes on to describe how the various practical difficulties preventing the application of SEA were overcome to the point where it could be used as a practical tool. Finally, a case history is presented in which an experimental SEA model is used to determine input (exciting) powers, power flow and sensitivity of the noise to parameter changes. In addition, the use of an optimization algorithm for driving the noise level down to a target spectrum within parameter and weight constraints is also demonstrated.

4:45

4pSA8. Professor Richards after his second retirement. Joseph. M. Cuschieri (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

Professor Richards, or Prof, as he liked to be called, had just come back from Loughborough University, where he was the Vice-Chancellor. This was his second retirement. While searching for an appropriate research area, he identified machinery noise as an important issue. He formed the machinery noise group in 1975, which I joined in 1980. At the time the group had seven members. The main emphasis of the work was in the prediction and control of noise from impact machines—punch presses, drop hammers, etc. Prof wanted to take a new look at the parameters that control impact noise. A major contribution was the energy accountancy concept. Prof was a constant source of new and fresh ideas, he would question any result or conclusion, and always contributed new insight to the problem and the results. Prof certainly had a unique approach to get to the root of an issue. I was fortunate that I kept in contact with Prof after I came to the US, as a fresh Ph.D., to join the faculty at Florida Atlantic University. His enthusiasm, inquisitiveness, and broad experience were invaluable and had a significant influence on my career and I am sure that of others.

Session 4pSC

Speech Communication and Psychological and Physiological Acoustics: Pediatric Cochlear Implants: Recent Studies of Speech Perception and Production

David B. Pisoni, Chair

Psychology Department, Speech Research Laboratory, Indiana University, Bloomington, Indiana 47405

Chair's Introduction—1:00

Invited Papers

1:05

4pSC1. Pediatric cochlear implantation: A review. Richard T. Miyamoto (Dept. of Otolaryngol., Indiana Univ., Indianapolis, IN 46202)

The field of pediatric cochlear implantation has changed tremendously since the early 1980s when single-channel cochlear implants were first provided to postlingually deafened children in the United States. Although implantation in children was at first restricted to older children with postlingually acquired hearing loss, it is now common for prelingually deafened children as young as two years of age to receive a cochlear implant. Furthermore, the introduction of multichannel cochlear implants and improved signal processing strategies have resulted in far greater speech perception benefits than originally thought possible. This presentation will review the major developments in pediatric implantation and summarize the results to date. [Work supported by NIDCD Grant Nos. DC00064 and DC00423.]

1:30

4pSC2. Long-term performance of prelinguistically deaf children using the Nucleus 22 channel cochlear implant. Steven J. Staller and Anne L. Beiter (Cochlear Corp., 61 Inverness Dr. E., 200, Englewood, CO 80112)

A sample of 178 children have been followed annually for the past five years, to assess the development of speech perception abilities in early deafened children using a multichannel cochlear implant. A subset of these children who acquired deafness prior to the acquisition of language (prelinguistic) have shown continued improvement on a hierarchy of pediatric speech perception tests throughout the follow-up period. Implantation at an early age appears to improve the prognosis for development of more difficult speech perception skills. In addition, there appears to be a positive relationship between the emphasis on oral listening skills and the rate of acquisition of speech perception abilities. Throughout the duration of the longitudinal study, several new speech processing strategies have been introduced for the Nucleus implant system. The most recent (SPEAK) is an adaptive strategy that varies the rate of stimulation and the number of electrode is stimulated according to the frequency characteristics of the speech signal. SPEAK attempts to represent the dynamic nature of speech and provides improved spectral detail by virtue of 22 closely spaced electrodes. A sample of 34 prelinguistically deafened children demonstrated significant improvements in speech perception within six months of conversion to the new processing strategy.

1:55

4pSC3. Clinical investigation of the CLARION cochlear implant in children. Mary Joe Osberger, Sue Zimmerman-Phillips (Adv. Bionics Corp., 12740 San Fernando Rd., Sylmar, CA 91342), and Sigfrid Soli (House Ear Inst., Los Angeles, CA 90057)

This presentation will summarize the design of the CLARION pediatric clinical trial and present initial findings on the subjects' performance with the device. A repeated measures design is used to compare each subject's preoperative performance with hearing aids to their postoperative implant performance on a battery of outcome measures. Six categories of audiologic outcome measures are administered to the subjects to achieve study end points that reflect the range of benefit predicted to occur in pediatric implant users over time. Postoperative evaluations with the CLARION are performed at 3, 6, and 12 months following initial stimulation, and semiannually thereafter. Initial findings comparing subjects' preoperative performance to their 3-month postoperative performance with the CLARION reveal improvements on all speech perception outcome measures. Performance on the outcome measures, however, is confounded by the limited cognitive and linguistic skills of some of the subjects, especially in the youngest (2- and 3-year-old) children. Information will be presented on issues inherent in the design and implementation of a large multicenter study of this nature.

2:20

4pSC4. Open set speech recognition in congenitally deaf children using a multichannel cochlear prosthesis. Susan Waltzman (Dept. of Otolaryngol., NYU School of Medicine, 550 First Ave., New York, NY 10016)

Thirty-eight congenitally deaf children received the Nucleus multichannel cochlear prosthesis at NYU Medical Center prior to age five. Preoperative and postoperative evaluations included pure tone audiometry under earphones, warble tone audiometry in the sound field (amplification system preoperatively and implant postoperatively), and open set measures of word/sentence recognition. Tests administered included, but were not limited to, GASP words and sentences, PBK words (scored as words and phonemes) and Indiana

phrases and were administered preoperatively and 6, 12, 18, 24, 36, 48, and 60 months postoperatively. Results indicated a significant improvement over time in the ability of the children to perceive words and sentences in the implant only condition. Percent correct scores improved with length of usage and increasing age, reflecting increased ability of the young children to perform the necessary tasks, in addition to developing speech recognition abilities. After using the device for two or more years the children had varying degrees of open set speech perception. These results demonstrate significant improvement in open set speech perception in congenitally deaf children implanted below the age of five.

2:45

4pSC5. Lexical discrimination and age at implantation. Karen Iler Kirk (Dept. of Otolaryngol., Indiana Univ., Indianapolis, IN 46202)

Recent research suggests that prelingually deafened children implanted at an early age may obtain greater speech perception and language benefits than those children implanted at a later age. This investigation examined word recognition and lexical discrimination in pediatric cochlear implant users as a function of age at implantation. Two groups of prelingually deafened children who used the Nucleus multichannel cochlear implant participated in this study. The first group contained children who received their device between the ages of 2–5 years, and the second contained children who were implanted between the ages of 6–9 years. Speech perception performance was evaluated using a new measure, the Lexical Neighborhood Test, a traditional measure of word recognition, the PBK, and a measure of receptive language abilities (the PPVT or the Reynell Developmental Language Scales). Children implanted at a younger age had better word recognition performance and were better able to identify lexically difficult words (i.e., those that occur infrequently and have many other phonemically similar words with which they can be confused). Correlations with vocabulary and language performance will also be reported. [Work supported by NIDCD DC00064.]

3:10–3:20 Break

3:20

4pSC6. Speech production and language development in pediatric cochlear implant users. Mario A. Svirsky (Dept. of Otolaryngol., Indiana Univ., Indianapolis, IN 46202)

Cochlear implantation has been very successful in postlingually deafened children. Implants provide them with substantial levels of speech reception, which in turn allows them to achieve better intelligibility, speech production, and English language proficiency. Congenitally or prelingually deafened children, while less successful than the aforementioned group, have also shown benefit in speech production and language measures. Performance varies greatly in this group, but there appears to be a trend towards better results when implantation is performed earlier in life. This presentation will analyze measures of speech production and language ability in children deafened before age 3 (who comprise the vast majority of pediatric cochlear implant candidates) as well as in children who are later deafened. Individual and group data will be discussed in view of the subjects' communication mode (oral or total communication), age at deafening, age at implant, and years of experience with the implant. [Work supported by NIDCD Grant Nos. DC00064 and DC00423.]

3:45

4pSC7. Speech production and use of sign by young cochlear implant users. Nancy Tye-Murray (Central Inst. for the Deaf, 909 S. Taylor, St. Louis, MO 63110), Linda Spencer (Univ. of Iowa Hospitals, Iowa City, IA 52242), and Bruce Tomblin (Univ. of Iowa, Iowa City, IA 52242)

The use of sign and the emergence of speech skills by children who are deaf and use a cochlear implant were examined. Twenty-five children who had an average of 43 months of experience with a Nucleus cochlear implant and who are educated in a simultaneous communication environment engaged in spontaneous conversation and completed speech production and audiological tests. During spontaneous conversation, the children used voice and sign to express whole words in their conversations 70% of the time, voice only 21%, and sign only 9% of the time, suggesting that use of a cochlear implant does not result in elimination of signing. Children demonstrated a wide range of intelligibility. Pearson correlations indicated that children who had better intelligibility were most likely to use voice only, while children who had poor intelligibility were most likely to use sign only. Analysis of the children's consonant production over time suggested that fricative production emerges relatively early; but follows emergence of other speech skills, including the ability to produce nasal consonants. Measures of consonant production were significantly correlated with measures of consonant perception. This suggests a close link between children's ability to produce speech and perceive speech.

4:10

4pSC8. Acoustic characteristics of speech in French children using multichannel cochlear implants. Emily A. Tobey (UTD-Callier Ctr., 1966 Inwood Rd., Dallas, TX 75235), Alain Uziel, Martine Sillon, Adrienne Vieu, and Francoise Artieres-Reuillard (Univ. of Montpellier, ORL, Montpellier, France)

Acoustic characteristics of speech production produced with and without auditory feedback from a multichannel cochlear implant was examined in eleven French speaking children with profound hearing losses. Subjects produced five repetitions of stimuli designed to contrast voice-onset times and vowel formant frequencies. Samples were collected with the implant turned on and after a 10-min period of the implant turned off. Stimuli were audio recorded, low-pass filtered, and digitized at a 10-kHz rate. VOT's and formant frequencies were measured using Cspeech software. VOTs for the bilabial cognates, /b/ and /p/, revealed considerable overlap in temporal values when no auditory feedback was available. During conditions providing auditory feedback, VOTs shifted to more nearly normal values with minimal overlap. Vowel formant frequencies also shifted as a function of implant status; however, the patterns of shift differed across subjects and across feedback conditions. Data suggest pediatric users of cochlear implants are able to use auditory feedback to adjust their speech production; however, the relationships between auditory conditions and speech acoustics appear complex. [Work supported by the Ministere de la Sante and a visiting research scholar award from the University of Montpellier.]

4:35

4pSC9. Using electrically evoked auditory potentials in the clinical management of pediatric cochlear implant users.
Carolyn J. Brown (Dept. of Otolaryngol.–Head and Neck Surgery, Univ. of Iowa Hospitals and Clinics, 200 Hawkins Dr., Iowa City, IA 52242)

For many young and/or prelingually deaf children, programming the speech processor of the cochlear implant and identifying malfunctioning electrodes or internal device failures can be difficult. Electrophysiologic recording techniques can be used to address both issues. The electrically evoked auditory brain-stem response (EABR) can be used to assist in device programming. Data describing the relationship between EABR threshold and behavioral measures of threshold and maximum comfort level for a group of 26 Nucleus cochlear implant users will be reviewed. Electrophysiologic techniques can also be used to record the stimulus artifact associated with activation of the Nucleus device. This response has been called the average electrode voltage (AEV) and has proven to be useful in identifying cases where there is total failure of the internal components of the Nucleus cochlear implant as well as diagnosing malfunction of individual electrodes. Normative data collected using three different stimulation modes from a group of 20 Nucleus cochlear implant users will be presented., Examples of commonly observed abnormal AEV recordings will be described and compared with AEVs recorded using an internal Nucleus electrode array submerged in a saline tank. [Work supported by NIH and the Iowa Lions Foundation.]

THURSDAY AFTERNOON, 16 MAY 1996 CELEBRATION B, 1:30 TO 4:20 P.M.

Session 4pUW

Underwater Acoustics: Signal Processing and General Topics

William M. Carey, Chair
DARPA/MSTO, 3701 North Fairfax Drive, Arlington, Virginia 22203

Chair's Introduction—1:30

Contributed Papers

1:35

4pUW1. Active matched-field tracking.
Homer Bucker and Paul Baxley (Code 541, RDTE Div. NCCOSC, San Diego, CA 92106)

The principal task of a sonar system is to generate possible tracks of an acoustic source. Matched-field tracking [Bucker, J. Acoust. Soc. Am. **96**, 3809–3811 (1994)] is a signal processing algorithm for this purpose that has minimum operator input. In this presentation, the method is applied to active sonar systems by extending the definition of covariance matrix elements. A sample simulation will be shown where operator-independent tracks are obtained at different noise levels.

1:50

4pUW2. Robust nonadaptive matched-field beamforming in an uncertain ocean environment.
Kerem Harmanci and Jeffrey L. Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., Box 90291, Durham, NC 27708-0291)

Matched-field processing techniques can be both ambiguity-prone and sensitive to errors in the assumed environmental conditions. Although greater robustness to environmental variability can be obtained by the use of data adaptive methods, such as the MV-EPC beamformer [J. L. Krolik, J. Acoust. Soc. Am. **92**, 1408–1419 (1992)] or joint estimation of source and channel parameters [D. F. Gingras and P. Gersoft, J. Acoust. Soc. Am. **97**, 3589–3598 (1995)], these strategies require higher signal-to-noise ratios. The methods presented here concern the design of robust nonadaptive beamformer weights with sidelobe levels which are less signal-to-noise ratio dependent. As with Chebyshev filters, the weights are designed to minimize the maximum magnitude-squared sidelobe level. Robustness to environmental mismatch is achieved by defining the sidelobe level as the magnitude-squared response averaged over an ensemble of environmental conditions. The two algorithms proposed are a fast iterative constrained minimum variance method and the use of a minimax quadratic programming technique. For an uncertain Mediterranean environment, an improvement in the maximum average sidelobe level of as much as 3 dB over the Bartlett beamformer is achieved. [Work supported by NRaD/ONR.]

2:05

4pUW3. The determination of optimum array lengths based on signal coherence in deep and shallow water.
William M. Carey (ARPA, 3701 N. Fairfax Dr., Arlington, VA 22203) and Peter Cable (BBN Systems and Technol., 1300 N. 17th St., Arlington, VA 22209)

Experimental measurements of signal coherence and array signal gain are reviewed for both deep and shallow water sound channels. The signal gain is related to single path or modal coherence lengths through well-known relationships in the statistical theory of antennas. Signal gain mea-

surements in the transverse and longitudinal directions are proffered as the optimum measures of coherence lengths for both broadband and narrow-band signals. Using this technique measurements (<1 kHz) are presented that show for the deep water cases lengths on the order of 300 wavelengths can be achieved while in the downward refraction conditions of a shallow water waveguide lengths between 30 and 100 wavelengths are realized. The measurement of broadband and narrow-band coherence and correlation functions are discussed with emphasis on the role of partly coherent noise backgrounds and multipath interference effects as well as averaging constraints. These results are interpreted with coherence models based on sound scattering from the volume and boundaries of the waveguide. The requirements for the numerical modeling of the signal coherence are presented.

2:20

4pUW4. Conjugate mirror array performance in the presence of surface and bottom roughness. Terry E. Ewart, Daniel Rouseff, and Darrell Jackson (Appl. Phys. Lab. and School of Oceanogr., Univ. of Washington, Seattle, WA 98105)

Dowling and Jackson [J. Acoust. Soc. Am. **91**, 3257–3277 (1992)] provided a study of the temporal/spatial effects of an internal wave field on phase conjugate arrays. In the present work the effects of interface roughness on phase conjugate arrays are simulated. The stochastic surface wave field is generated to include realizations of the sea surface height that evolve in time. The time variability of the surface also includes the effects of the acoustic travel time, so that each realization is "as viewed" in the frame of the acoustic wave from source-array-to-receiving-array and the return after phase conjugation., The acoustic wave propagation is accomplished using a parabolic equation marching algorithm that includes this random rough surface. Bottom penetration and absorption are included. Coupling of the surface wave field with the bottom/sub-bottom randomness can produce an increased degradation of the return field. Thus even if the processing time to produce the conjugate field approached zero, degradation of the conjugate array focus will occur. A discussion of the relevance of this work to conjugate array applications will be presented. [Work supported by ONR.]

2:35

4pUW5. Model-based processing of broadband sources in noisy shallow ocean environments. James V. Candy (Lawrence Livermore Natl. Lab., Univ. of California, P.O. Box 808, L-495, Livermore, CA 94550) and Edmund J. Sullivan (Naval Undersea Warfare Ctr., Newport, RI 02841)

Broadband acoustic sources propagating in a hostile (noisy) ocean complicate the analysis of received acoustic data considerably. Normal-mode models are reasonable propagators for use in a shallow ocean environments. Our previous work developed the Gauss–Markov representation of the narrow-band normal-mode model and then recently extended it to the broadband case [Candy and Sullivan, J. Acoust. Soc. Am. Suppl. 1 **98**, (1995)]. In this paper the design and trade-offs of the processor are investigated when applied to both synthesized and experimental data in order to construct the desired broadband processor. The enhancement of broadband acoustic pressure-field measurements along with the estimation of underlying model functions using a vertical array is discussed. The model-based approach is developed and implemented for a broadband source using a normal mode propagation model. The structure of the processor is of interest, since it has an implied parallel structure due to the propagation physics, while the optimal estimation solution implies a "full" structure. The apparent discrepancy is resolved and the processor is implemented using a "bank" of narrow-band model-based processors. The resulting broadband pressure-field estimates are quite reasonable for both synthesized and experimental data.

2:50–3:05 Break

3:05

4pUW6. Travel time variability and localization accuracy for global scale monitoring of underwater acoustic events. Ted Farrell (BBN Systems and Technologies, 1300 N. 17th St., Arlington, VA 22209) and Kevin D. LePage (BBN Acoustic Technologies, Cambridge, MA 02138)

Large amplitude underwater acoustic signals, such as those produced by earthquakes and nuclear tests, can be monitored on a global scale using a limited network of underwater acoustic sensors. The design and operation of such a network requires an understanding of the limits on localization accuracy due to sound speed fluctuations in the ocean. For localization based on time of flight, the variance of the arrival time of the individual modes may be estimated by a range and depth integral over the sound-speed fluctuation statistics of the water column. Results obtained for travel time variance using this approach are compared to simpler measures such as the variance of the sound axis slowness integrated over a geodesic path. These results are in turn utilized to estimate the localization areas of uncertainty for a variety of source and receiver scenarios. All results are based on measured sound-speed statistics obtained from various ocean databases. [Work supported by DOE and Air Force Phillips Laboratory.]

3:20

4pUW7. An empirical fractal model for corona discharges in salt water. J. C. Espinosa, H. M. Jones, A. M. Gleeson, and R. L. Rogers (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A corona discharge in water and a concomitant acoustic pulse is produced when a high electric field is applied across a pair of electrodes. The corona structure consists of many branching plasma fingers. The acoustic signal is produced by the formation and collapse of a vapor bubble; however, the details of this process are unclear. Furthermore, the relation between the acoustic and fractal stages is unknown. Voltage and current waveforms, obtained previously, are accurately described by a fractal model. High-speed photographs of discharges have been taken to further investigate the nature of corona discharges and elucidate the relationship between the two stages. From these experiments, an empirical model is under development that incorporates both the fractal and acoustic aspects of corona discharges. The most recent results of this effort will be presented. [Work supported by the Office of Naval Research.]

3:35

4pUW8. Cylindrical bubble evolution and acoustic signature through the arc phase of an electrical discharge. David L. Fisher and Robert L. Rogers (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 71713-8029)

The arc phase of an electrical discharge in salt water is investigated using a 1-D nonlinear fluid model in cylindrical (r) coordinates. For most electrode geometries, the arc phase is more accurately modeled using a cylindrical geometry compared to the usual and simpler to implement spherical model. This is due to the fact that until the bubble radius is comparable to the distance between electrodes, the preferred geometry is cylindrical. Both the bubbles external (water) and internal (dissociated

water and plasma) are discretized and modeled with nonlinear fluid equations. The model includes the energy flow from the capacitor, into the plasma arc through its resistivity, and then finally into the hydroacoustic pulse. Simulation results will be compared to experiments. Also the efficiency of various electrode configurations will be investigated. [Work supported by the Office of Naval Research.]

3:50

4pUW9. Underwater turbulence injection system for investigation of acoustic propagation in a randomly fluctuating medium. Barry J. Doust, Kenneth E. Gilbert, and Ralph R. Goodman (Appl. Res. Lab. and the Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

A preliminary investigation using cold water from melting ice to induce sound-speed fluctuations has shown significant effects on both one-way and two-way propagation. Although the melting ice approach is simple and effective, it is not easily controlled. The purpose of the present research is to develop a method for generating controlled sound-speed fluctuations by creating a thermal field with stable and repeatable statistics. The work has led to a novel cold water injection system for generating thermal fluctuations in an experimental tank. A relatively simple device has been designed for producing a range of turbulence spectra by varying the flow rate and orifice sizes. This paper will focus on the characteristics of the cold water injection system and will discuss its advantages over systems that use, for example, electric heaters or hot water injection to create the thermal fluctuations. Results from preliminary acoustic tests will be discussed. [Work supported by the Office of Naval Research and the Applied Research Laboratory.]

4:05

4pUW10. Observation of low-frequency wave generation in a bubble layer. Alexander M. Sutin, Irina A. Soustova, Alexander I. Matveyev, Andrey I. Potapov (Inst. of Appl. Phys., Russian Acad. Sci., 46 Ulyanov str., Nizhny Novgorod, 603600, Russia), and Lev. A. Ostrovsky (Inst. of Appl. Phys. and Univ. of Colorado, CIRES/NOAA Environmental Technol. Lab., Boulder, CO 80303)

The difference-frequency sound generation as a result of the interaction of two high-frequency harmonic waves in a bubble layer in water are investigated experimentally and theoretically. As it was shown before, the use of the layer resonance can increase the efficiency of the nonlinear transformation of the signal. Here the data of the experiment with a bubble layer of the thickness of about 10 cm in the anechoic tank are presented. One of the incident (primary) wave frequencies was 60 kHz while the another varied from 59 to 50 kHz, thus providing the low-frequency signal in the range of 1 to 10 kHz. Due to the first-mode layer resonance, this secondary signal had a pronounced maximum at the frequency of 2.2 kHz (while the primary wave was in resonance with the bubbles of the radius about 45 μm, and no resonance in the layer was observed for it). The bubble volume ratio from these experiments was estimated to be about 1.4×10^{-4}. A theory was also developed for this type of interaction which described well the experimental results. [Work was supported by the Russian Foundation of Fundamental Research Grant No. N0-93-05-8074 and by the Science Opportunity Program of the Office of Naval Research.]

THURSDAY AFTERNOON, 16 MAY 1996 — NATIONAL PARKS, 1:30 TO 2:45 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

U. S. Technical Advisory Group (TAG) Meetings for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

T. A. Frank, Chair S3
Pennsylvania State University, Speech and Hearing Clinic, 110 Moore Building, University Park, Pennsylvania 16802

R. F. Burkhard, Vice Chair S3
Hearing Research Laboratoriy, State University of New York at Buffalo, 215 Parker Hall, Buffalo, New York 14214

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U. S. CERL, P.O. Box 4005, Champaign, Illinois 61820

H. E. von Gierke, Vice Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 108/SC4, Human Exposure to Mechanical Vibration and Shock
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U. S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged. The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock, will also be discussed. The Chairs of the U. S. Technical Advisory Groups for ISO/TC 43 (P.D. Schomer), IEC/TC 29 (V. Nedzelnitsky), and ISO/TC 108/SC4 (H. E. von Gierke) will report on current activities of these international Technical Committees and Subcommittees, including the meetings of ISO/TC 43 and IEC/TC 29 held in Pretoria, South Africa, in February 1996.

Scope of S3. Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical shock and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance, and comfort.

THURSDAY AFTERNOON, 16 MAY 1996 NATIONAL PARKS, 3:00 TO 5:00 P.M

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

U. S. Technical Advisory Group for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics

G. S. K. Wong, Chair S1
Institute for National Measurement Standards (INMS), National Research Council, Ottawa, Ontario K1A 0R6, Canada

R. W. Krug, Vice Chair S1
Cirrus Research, Inc., 6423 West North Avenue, Suite 170, Wauwatosa, Wisconsin 53213

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U. S. CERL, P. O. Box 4005, Champaign, Illinois 61820

H. E. von Gierke, Vice Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U. S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899

Standards Committee S1 on Acoustics. Working group chairs will report on their preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged. The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, will also be discussed. The chairs of the respective U. S. Technical Advisory Groups for ISO/TC 43 (P.D. Schomer) and IEC/TC 29 (V. Nedzelnitsky) will report on current activities of these international Technical Committees, including the meetings of ISO/TC 43 and IEC/TC 29 held in Pretoria, South Africa, in February 1996.

Scope of S1. Standards, specifications, methods of measurement, and test and terminology in the field of physical acoustics including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Session 5aAB

Animal Bioacoustics: Effects of Noise on Animals

Larry L. Pater, Chair
USA-CERL, P.O. Box 9005, 2902 Newmark Drive, Champaign, Illinois 61826-9005

Chair's Introduction—8:00

Invited Papers

8:05

5aAB1. The effect of roadway traffic noise on territory selection by Golden-cheeked Warblers. Robert H. Benson (Ctr. for Bioacoustics, Texas A&M Univ., College Station, TX 77843-3367)

Widespread concern that environmental noise produced by humans can negatively affect animal populations is reflected in a growing body of literature. This study evaluates the possible effects of roadway traffic noise on territory selection by the endangered Golden-cheeked Warbler. Seventy-eight listening posts were established randomly in a 212 ha study area in central Texas. Noise exposure at each post was estimated using a noise simulation model based on traffic counts. The presence or absence of warblers at each post was determined by field observations. Data were analyzed to determine if a correlation existed between the estimated noise exposure and the occurrence of warblers at a listening post. Exposure in $L_{eq(h)}$ ranged from 29.7–58.6 dB. Warblers were detected at 30 of the 78 listening posts. When the 78 posts were divided into high-noise and low-noise groups, there was no significant difference between the occurrence of warblers in the groups. Logistic regression failed to reveal a significant correlation between the occurrence of warblers at listening posts and the exposure to noise. It is concluded that, within the range of noise exposures considered in this study, Golden-cheeked Warblers do not select territories based on exposure to roadway traffic noise.

8:25

5aAB2. USAF monitors for recording aircraft noise received by animals. Michael Carter (Armstrong Lab., Wright-Patterson AFB, OH 45433) and Robert Kull (Armstrong Lab., Brooks AFB, TX 78235)

Although researchers have studied the effects of aircraft noise on wild and domestic animals for many years, accurate descriptions of the noise levels received by the animals in many of the studies was not verified. Recent technology has allowed for the miniaturization of much of the hardware for noise monitoring devices, making it feasible to build a noise monitor small enough to fit on a large animal collar. An animal noise monitor (ANM) was designed and built to capture A- and C-weighted noise levels above a programmable threshold, distinguishing aircraft noise from other sound sources. The device, weighing only 350 g, also captures onset rate, *Leq*, and gross movements of the animal via accelerometers after a noise event, while fitting on a large animal collar. The ANMs were also designed to function as stand-alone, weatherproof units for up to 6 months. The ANMs have already been field tested under natural weather conditions at stationary locations by Peregrine Falcon aeries around Fairbanks, Alaska. The ANMs will be tested on penned animals to determine their reliability and accuracy.

8:45

5aAB3. Performance of desert kit foxes, *Vulpes macrotis arsipus*, on acoustic detection trials during simulated jet aircraft overflights. Ann E. Bowles, Scott Yaeger (Hubbs-Sea World Res. Inst., 2595 Ingraham St., San Diego, CA 92109), B. Andrew Kugler (BBN Systems and Technologies Corp., Canoga Park, CA 91303), and Richard Golightly (Humboldt State Univ., Arcata, CA)

5a FRI. AM

Two male and female desert kit foxes were exposed to simulated prey and predator sounds in the presence of aircraft noise to determine detection rates. They were tested in a 9.75×9.15-m outdoor pen in the dark. Five feeding stations were equipped with speakers to play scratching sounds (prey noise) and a single speaker outside the pen projected stealthy footsteps (predator noise). Foxes approached feeding stations projecting prey noise to receive a reward; they spontaneously avoided predator noise. Foxes were exposedto 980 simulated low-altitude jet overflights in the range from 65 to 95 dB ASEL. Stimulus sounds were played for 1 min at the limit of detection; the longest simulated overflight was 45 s. Fox performance was compared with and without simulated aircraft noise (309 and 110 trials, respectively). Foxes exhibited flight reactions or dropped flat during initial exposures. Within 1–2 series of exposures, these responses habituated, although milder reactions persisted. Foxes identified the correct feeding station 63%–71% of the time (chance rate=20%) in silence versus 24%–30% in noise. They detected predator sounds on 81%–83% of trials, regardless of noise. The results suggest that foxes listened for potential predators preferentially in the presence of aircraft noise. [Work supported by USAF NSBIT Program, Contract F33615-90-D-0653.]

9:05

5aAB4. Effects of underwater sound on hair cells of the inner ear and lateral line of the oscar (*Astronotus ocellatus*). Mardi C. Hastings (Dept. of Mech. Eng., Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210), Arthur N. Popper (Univ. of Maryland, College Park, MD 20742), James J. Finneran (Ohio State Univ., Columbus, OH 43210), and Pamela J. Lanford (Univ. of Maryland, College Park, MD 20742)

Fifty-nine oscars (*Astronotus ocellatus*) were subjected to pure tones at 60 or 300 Hz in order to determine the effects of sound at levels typical of man-made sources on the sensory epithelia of the ear and the lateral line. Sounds varied in duty cycle (20% or continuous) and sound-pressure level (100, 140, or 180 dB *re*: 1 μPa). Fish were allowed to survive for one or four days post-treatment. The study also included five control animals that were handled in the same way as the test animals, but were not exposed to any sound. Tissue was then evaluated using scanning electron microscopy to assess the presence or absence of ciliary bundles on the sensory hair cells in each of the otic endorgans and the lateral line. Damage was found in four of five fish stimulated with 300-Hz continuous tones at 180 dB and allowed to survive for four days. It was limited to small regions of the striola of the utricle and lagena. No damage was observed in fish that had been allowed to survive for one day post-stimulation, suggesting that physical signs of damage may develop slowly after exposure. [Work supported by ONR.]

9:25

5aAB5. The impact of impulsive noise on bald eagles at Aberdeen Proving Ground, Maryland. William A. Russell, Jr., Nelson D. Lewis (Environ. Noise Program, U.S. Army Ctr. for Health Promotion and Preventive Medicine, ATTN: MCHB-DP-B, Aberdeen Proving Ground, MD 21010-5422), and Bryan T. Brown (SWCA, Inc. Environ. Consultants, Flagstaff, AZ 86001)

The U.S. Army Aberdeen Proving Ground supports one of the largest bald eagle concentrations on the Northern Chesapeake Bay (Buehler *et al.*, 1987). The testing of large caliber weapons and detonation of large explosive charges at U.S. Army Aberdeen Proving Ground is creating significant noise and producing concerns on the potential effects on bald eagles. The population using Aberdeen consists of approximately 15 nesting pairs and up to 100 wintering or nonbreeding residents. Systematic observations on potential influence of noise on bald eagles were made from November 1993 through December 1995. We detected no response in a high percentage of nesting and roosting eagles exposed to numerous types of noise at varying levels of intensity.

9:45

5aAB6. Discussion and demonstration of PC CITASAN. Paul A. Sharp (2610 Seventh St., Wright Patterson AFB, OH 45433-7901)

PC CITASAN is a software program that contains a database of over 4500 references to scientific literature on the impact of aircraft noise and sonic booms on humans, animals, structures, and noise modeling. It contains bibliographic data and a suitability rating and a controversiality rating for each article. Some articles also have an abstract and an independent review. The data were updated as of Nov 1994. The PC CITASAN database can be searched using four different pre-defined screens, one for each impact category. The search can be global, by author, by key words in the title, or by year of publication. The program was developed for use by USAF personnel preparing environmental assessment documents. However, it has broad application to other government, civilian, academic, and commercial organizations. The program was originally written to operate under the UNIX operating system using ORACLE as the database. It has been completely re-written to run under the Windows environment on a PC as a stand alone system and does not require a dedicated database. It will be distributed on a CD-ROM with a hardcopy user's manual.

10:05–10:20 Break

10:20

5aAB7. Field methods for measuring auditory function in wild animals. Ann E. Bowles (Hubbs–Sea World Res. Inst., 2595 Ingraham St., San Diego, CA 92109)

Laboratory characterization of auditory sensitivity in wild vertebrates is typically based on measurements from a small number of relatively healthy and young individuals. The auditory sensitivities of animals in *populations* are likely to vary greatly; individuals will vary in sensitivity at birth, and will experience diminished capacity due to injuries, noise exposure, age, starvation, reproductive stresses, parasites, etc. Although disabilities might be fatal for hearing-dependent species, such as nocturnal animals, healthy individuals with profound loss are sometimes found. Hypotheses about the adaptive value of hearing cannot be addressed without

techniques to measure animal hearing quickly and accurately in field situations. While conditioning ("behavioral") studies are the "gold standard" against which other techniques are evaluated, they cannot be used easily in the field. Examples of other techniques will be given and compared with behavioral measures for the same or similar species. These are (1) reflex modification (startle inhibition) used on 4 kit foxes (*Vulpes macrotis*) and 4 Merriam's kangaroo rats (*Dipodomys merriami*), (2) auditory brain-stem responses used on 16 kangaroo rats and 12 desert tortoises (*Gopherus agassizii*), and (3) envelope-following responses used on a desert tortoise and a harbor seal (*Phoca vitulina*).

10:40

5aAB8. Response of nonbreeding spotted owls to helicopter and chain saw noise. Teryl G. Grubb, David K. Delaney (Rocky Mountain Forest and Range Experiment Station, USDA Forest Service, Flagstaff, AZ), M. Hildegard Reiser (Holloman Air Force Base, NM), Paul Beier (Northern Arizona Univ., Flagstaff, AZ), and Larry L. Pater (U.S. Army Construction Eng. Res. Labs., Champaign, IL)

Mexican spotted owls (*Strix occidentalis lucida*) were exposed to two types of noisy stimuli: very low altitude helicopter overflights and chain saw activity. The stimuli were presented to the owls at several distances and noise levels during daylight hours in July through September 1995. Responses were characterized into two categories, "alert" and "flight." "Alert" responses to low altitude helicopter overflights were typical, but "flight" responses occurred for only a small percentage of the trials. Chain saw presentations at similar distances elicited "flight" responses much more frequently. Consideration was given during data analysis to the owls' hearing range and sensitivity. [Research supported by U.S. Air Force.]

Contributed Papers

11:00

5aAB9. Sensitization and habituation to underwater sound by captive pinnipeds. David Kastak and Ronald J. Schusterman (Long Marine Lab., Univ. of California, Santa Cruz, CA 95060)

Responsiveness of three species of pinniped to two types of brief (5-s duration) underwater sound was investigated. The experimental stimulus was a broadband pulsed sound (peak levels of 110–135 dB *re:* 1 μPa). The control stimulus was a frequency sweep (rms level of 125–130 dB *re:* 1 μPa). Reactions were classified as avoidance or approach. Two California sea lions and a harbor seal showed no avoidance reaction to either stimulus. A northern elephant seal showed a powerful avoidance response to the pulsed sound and showed little reaction to the frequency sweep. The sea lions and harbor seal clearly showed habituation to both sounds. However, rather than showing habituation, the elephant seal showed sensitization to the experimental stimulus, as well as to the entire experimental setup. That is, as the experiment progressed, avoidance response to the signal was faster and duration of post-stimulus haulout behavior increased. In fact, following successive presentations of the pulsed sound, it was often not possible to coax the elephant seal back into the water. The results suggest that there are species-specific constraints on habituation to certain types of environmental stimuli.

11:15

5aAB10. Effects of sound and ultrasound on Zebra Mussels. Dimitri M. Donskoy (Davidson Lab., Stevens Inst. of Technol., Hoboken, NJ 07030), Michael Ludyanskiy (LONZA, Inc., Annandale, NJ 08801), and David A. Wright (Univ. of Maryland, Solomons, MD 20688)

The freshwater bivalve mollusk, *Dreissena polymorpha* – better known as the Zebra Mussel, since its introduction into the Great Lakes in 1985 has been quickly spreading throughout the waterways of both the United States and Canada causing serious economic and environmental consequences. The present study has been directed toward developing acoustic techniques to control Zebra Mussel infestation in various water intake and storage facilities. The effects of ultrasonic and hydrodynamic cavitation, low-frequency sound and vibration on various life stage mussels (from eggs and larvae to adults) have been studied. It was found that cavitation can be used as a control measure for Zebra Mussel veligers. The efficiency of the ultrasonic and hydrodynamic cavitation treatments were measured as an output energy rate to achieve 100% mortality. Laboratory experiments demonstrated the effective use of low-frequency waterborne sound to prevent Zebra Mussels from settling and translocating and to reduce reproduction abilities of mussels. [This work has been supported by Grant No. NA26RG0403-01 from the NOAA to the Stevens Institute of Technology, New Jersey Sea Grant College Program.]

Session 5aNS

Noise: Topics in Community Noise and Noise Control

Gerald C. Lauchle, Chair
Graduate Program in Acoustics, Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Contributed Papers

8:30

5aNS1. Problem areas in predicting community noise that affect accuracy. Frank H. Brittain (Bechtel Corp., 50 Beale St., San Francisco, CA 94105)

In theory, predicting community noise levels is relatively easy — once the needed information can be found. While the equations needed to formulate a noise prediction model are in the literature, unfortunately, they are spread between many references. Adequate information on noise levels of sources — either as sound-pressure or power levels — is often not available in the literature. Further, transmission loss data at low frequencies are not usually available. When one gets around these problems, another set of problems arises that is related to the accuracy of the models developed. These problems include source size and directivity, barrier and screening effects of equipment, reflections by equipment and buildings, piping and minor equipment, atmospheric and ground effects, and effectiveness of noise control barriers. Typical prediction technology is summarized. Problem areas that affect accuracy of noise predictions are identified and discussed.

8:45

5aNS2. Relationship between judgments of neighborhood noisiness and prevalence of annoyance. Sanford A. Fidell and Laura A. Silvati (Acoustic Technologies Div., BBN Corp., 21120 Vanowen St., Canoga Park, CA 91303-2853)

Dosage-response functions that relate the time-weighted daily average sound power of outdoor noise to the prevalence of a consequential degree of noise-induced annoyance in communities are often viewed as providing a basis for land use compatibility policy recommendations. Although such functions advance understanding of the rate of growth of annoyance with sound level and of related matters, they do not in themselves dictate any particular policy recommendations. It is therefore useful to consider how the growth of noise-induced annoyance is itself related to other nonacoustic measures of community response to noise exposure, such as the rate of growth of neighborhood noisiness judgments with sound level. A probabilistic model of community response to noise exposure developed by Green and Fidell (1991) is shown to provide a good account of a set of neighborhood noisiness judgments. The model is used to relate the growth functions for judgments of neighborhood noisiness and the prevalence of annoyance to one another.

9:00

5aNS3. Design and effectiveness of in-line tuning cables for quieting hydraulic power units. John M. Dodson, David R. Dowling, and Karl Grosh (Dept. of Mech. Eng. and Appl. Mechanics, Univ. of Michigan, Ann Arbor, MI 48109-2125)

The use of hydraulic units to power tools, lifts, and other industrial equipment in both indoor and outdoor settings is widespread. The noise radiated by these units and the hydraulic lines used to convey the fluid continues to be a problem. The dominant noise components arise from pressure fluctuations in the hydraulic fluid ocurring at harmonics of the pump vane or piston passage frequency. In this study, the effectiveness of in-line tuning cables (after a patent by Klees 1967) for quieting the harmonic noise from these hydraulic lines is determined. This approach, which combines ease of installation with relative economy, amounts to the insertion of a quarter-wavelength side branch into the acoustical circuit. Critical to effective design is the determination of the wave speeds in the flexible hosing. Theoretical predictions and measurements are applied to determine the wave speeds in the fluid-loaded hydraulic line. Multiple PVDF sensors measure wave speeds and the wave amplitude insertion loss achieved by the device. Experiments performed on a full-scale 60 h.p., 1500 p.s.i. unit will be compared to the theory. Sound pressure levels determine design effectiveness.

9:15

5aNS4. A study of noise reduction by antisound using bypass in ducts with flow. Zhichi Zhu, Rui Guo (Dept. of Eng. Mechanics, Tsinghua Univ., Beijing, 100084, People's Republic of China), and Rui Tien (Inner Mongolia Polytechnic Univ., Huhhot, People's Republic of China)

Noise reduction in ducts with flow is a very useful project. With common antisound methods, electronic instruments are required, so its application is restricted. By using bypass added to the main duct with flow, downstream noise reduction can also be obtained if suitable bypass parameters are selected: This simple method is completely in accord with antisound principle. But sound field analysis in this case is quite a complicated problem. The numerical and experimental research was completed by the authors. The numerical work included formulating acoustic relations and joint conditions and calculating the transmission loss for various noise sources. The experimental research on a duct with single bypass and outlet absorbing wedge under the conditions of different noise sources and flow velocities was carried out in a national lab. The measured data were in good agreement with the numerical results and considerable noise reduction was obtained in both dB(A) and 1/3 octave band. Finally, a spot test of noise reduction using double bypasses was carried out. It was shown that a noise reduction of 5 dB(A) was obtained even though the situation was not in optimization. [Work supported by NNSF of China.]

9:30

5aNS5. Quiet motor design. Cüneyt Öztürk, Emin Sönmez, Harun Açikgöz, and Birdal Göok (TEE A.Ş, R&D Dept., Davutpaşa, Litros Yolu 1, Topkapi 34020, Istanbul, Turkey)

Acoustic modeling of appliance motors leads the product designers to fully grasp all the technical features of the motor that cause noise. Utilizing complete analysis packages during prototype development provides the capabilities of existing numerical analysis software to enable the designer to visualize the influence of all mechanical, electromagnetic, and acoustic

features on the potential product. The purpose of this study was to develop a methodology to let the designer predict all parameters including mechanical resonances, electromagnetic forces, and aerodynamic and acoustic features that dominate noise spectrum of actual prototypes, during the development stage. This method enables the designer to be aware of all parameters that cause the increase of the audible noise and let them know the facts of using predictive design tools in an efficient and correct way on the basis of developed methodology. The result of this research is a design methodology that leads to the development of low noise products and efficient use of potential design tools.

FRIDAY MORNING, 17 MAY 1996 REGENCY C, 8:00 A.M. TO 12:10 P.M.

Session 5aPA

Physical Acoustics: Resonant Ultrasound Spectroscopy I

Albert Migliori, Chair
Los Alamos National Laboratories, MS-K764, P.O. Box 1663, Los Alamos, New Mexico 87544

Chair's Introduction—8:00

Invited Papers

8:10

5aPA1. Temperature variation of elasticity of α-quartz by the rectangular parallelepiped resonance method. Ichiro Ohno (Dept. of Earth Sciences, Ehime Univ., Bunkyou-cho 2-5, Matsuyama, 790-77 Japan)

The resonant trasound spectroscopy of a specimen of α-quartz single crystal was carried out up to 558 °C, just below the $\alpha-\beta$ transition ($_0$=573 °C), and all of the independent elastic constants were determined simultaneously by the rectangular parallelepiped resonance method. The results show that bulk moduli decrease largely toward T_0, while the shear moduli show only slight decreases. The constant, C_{14}, characteristic of trigonal crystals, decreases to just below T_0, but it seems not to vanish, but to remain a finite value, even at T_0. It is noted that the isotropic aggregate of α-quartz is a curious and scarce material, in the sense that its bulk modulus is smaller than the shear modulus, and Poisson's ratio is very small or even negative at temperatures higher than 450 °C. The effect of piezoelectricity of α-quartz on the theoretical frequencies was investigated in order to know whether the ordinary nonpiezoelectric theory gives considerably inaccurate resonance frequencies. It was shown that the nonpiezoelectric theory gives a frequency lower than the frequency given by piezoelectric theory, but the difference is 0.5% at most.

8:40

5aPA2. The determination of physical properties at high pressure using a high-temperature database from resonant ultrasound spectroscopy (RUS) measurements. Orson L. Anderson, Hyunchae Cynn, and Donald G. Isaak (Inst. of Geophys. and Planet. Phys., UCLA, Los Angles, CA 90095-1567)

Using the RUS technique called rectangular parallelepiped resonance, a large database on elastic constants and associated thermoelastic parameters extending from 300 K up to as high as 1800 K (P=0) has been established at UCLA. Temperature derivatives of the C_{ij}'s have been determined with considerable precision. It has been shown how the pressure derivatives of elastic constants $(\partial C_{ij}/\partial P)_T$ can be approximated from $(\partial C_{ij}/\partial T)_P$. Agreement with experiment is quite good in some cases. Extending this method to high pressure (>3 GPa) requires evaluation of the volume dependence of the parameter, $(\partial P/\partial T)_V=\alpha K_T$, where α is thermal expansivity and K_T is the isothermal bulk modulus. Again, this is done from temperature C_{ij} data, but it requires one datum on pressure, $K_0'=(\partial K_T/\partial P)_T\ (P=0)$. From the data, the temperature at which αK_T becomes independent of volume was predicted. The theory agrees well with experiment for all solids tested (NaCl, MgO, Al_2O_3, Mg_2SiO_4, CaO).

9:10

5aPA3. New pressure measurements of elasticity using the rectangular parallelepiped resonance method. Donald G. Isaak, O. L. Anderson, J. D. Carnes, and H. Cynn (Inst. of Geophys. and Planet. Phys., UCLA, Los Angeles, CA 90095-1567)

The rectangular parallelepiped resonance (RPR) method has proven to be an effective means to measure the adiabatic elastic moduli, C_{ij}, at very high temperatures. The extension of the RPR technique to elevated pressure is an attractive proposition because bonding problems associated with coupling between transducers and specimens are essentially eliminated. However, RPR data have not yet been reported at elevated pressure because of (1) the small frequency shifts observed over the accessible pressure range, and (2) complications in the data reduction scheme by which primary measurements of a resonance spectrum are interpreted in terms of the specimen C_{ij}'s. The experimental and analysis problems encountered when pressure dependences of modal frequencies are measured will be discussed, and ways in which these problems can be overcome will be presented. A new apparatus in which RPR measurements at elevated pressure have been made will be described. Initial RPR pressure data on several materials will be presented and compared to results obtained when using thermodynamic considerations that relate the temperature dependences of the resonant frequencies to their pressure dependences.

9:40

5aPA4. Resonant ultrasound spectroscopy and Raman measurements to examine cation order–disorder. Hyunchae Cynn, Orson L. Anderson, and Donald G. Isaak (Inst. of Geophys. and Planet. Phys., UCLA, Los Angeles, CA 90095-1567)

The elastic constants of natural $MgAl_2O_4$ spinel have been measured using resonant ultrasound spectroscopy at high temperature. Below 1000 K, the ultrasonic resonant frequencies of an ordered natural spinel change significantly after heat treatment. The Raman spectra for ordered natural and disordered spinels also differ below 1000 K but are similar at higher temperatures and after cooling to ambient temperature. These changes in ultrasonic resonance and Raman spectra of spinel can be associated with cation disordering at high temperature that may be quenched by cooling. The estimated inversion parameters from the relative intensities of the two A1g Raman modes are in very good agreement with estimates from other measurements. It is found that C_{11} and C_{12} decrease by 4% and 8%, respectively, with 20% inversion in spinel; C_{44} is less sensitive to cation disorder. These results imply that previous measurements of the adiabatic elastic constants of spinels at ambient conditions have been affected by the state of cation disorder of the specimen.

10:10–10:40 Break

Invited Poster Papers

Authors of papers 5aPA5 and 5aPA6 will be at their posters during the break.

5aPA5. Measurements for pressure and temperature dependencies of elastic moduli by the resonant sphere technique, RST. Isao Suzuki, Tomonori Fujio, Hiroshi Kikuchi, Hitoshi Oda (Dept. of Earth Sciences, Okayama Univ., Okayama, 700 Japan), and Ichiro Ohno (Ehime Univ., Matsuyama, 790-77 Japan)

The resonant sphere technique, RST, has been developed for measurements of elasticity and anelasticity of small crystal specimens. This method has advantages over other methods, especially in high-temperature and high-pressure measurements. New methods of data acquisition have made it convenient to measure resonance frequency at high temperatures (using a buffer rod), making a 1 K temperature interval measurement possible; this opens up many possibilities in the determination of physical properties of solids. Preliminary measurements of resonance frequency were performed by RST up to 100 MPa, which showed interference of vibration modes between the specimen and the pressure medium, even for helium gas as the pressure medium. This interference may become more serious at higher pressures or under liquid pressure. In order to evaluate such effects, the cavity resonance method was developed by Ohno in 1993, with a spherical shell structure with a spherical specimen at the center. This gives clear boundary conditions in the wave equations. Theoretical evaluation of the resonant frequency on such a system shows the necessity of correction to pressure derivatives of elastic moduli, even in measurements up to 100 MPa.

5aPA6. Simultaneous determination of elastic constants and asphericity of aspherical specimen by the resonant sphere technique. Hitoshi Oda (Dept. of Earth Sciences, Okayama Univ., Okayama, 700 Japan)

The resonant sphere technique (RST) is a method of resonant ultrasound spectroscopy developed to measure elastic constants of solids. In this method, resonant frequencies of a spherical specimen are measured and the elastic constants are determined by comparing the measured frequencies with theoretical ones that have been computed for a set of elastic constants. A perfect sphere is not always obtained, however; the specimen sometimes has small asphericity. In this case, the effect of asphericity on the resonant frequencies has to be corrected. Thus a method has been developed to determine simultaneously the elastic constants and asphericity of an aspherical specimen. When RST is employed for elasticity measurements of an ellipsoidal specimen, the difference between measured and computed resonant frequencies is expressed by $\delta\omega=\phi_i\epsilon_i+A_{ij}\delta C_{ij}$, where summation convention is assumed for repeated indices, and ϵ_i and δC_{ij} $(i,j=x,y,z)$ are asphericity of the ellipsoid and small corrections for a set of elastic constants, respectively. Since the coefficients ϕ_i and A_{ij} are known, the unknown coefficients ϵ_i and δC_{ij} can be determined by a least-squares method. Actual application will be reported for an ellipsoid of an olivine specimen with orthorhombic crystal symmetry.

Contributed Papers

10:40

5aPA7. Voigt stiffnesses and amplitude-dependent internal friction in monocrystal silicon. Hassel Ledbetter, Sudook Kim, Christopher Fortunko (NIST, 325 Broadway, Boulder, CO 80303), Paul Heyliger (Colorado State Univ., Fort Collins, CO 80523), and Mitsuru Tanaka (NRLM, Tsukuba, Ibaraki, 305 Japan)

The specimen consisted of a 1-cm-diam monocrystal silicon sphere. Voigt elastic stiffnesses C_{11}, C_{12}, C_{44} were determined from the macroscopic eigenvibration frequencies. The C_{ij} results agree closely with previous measurements by conventional acoustic methods. The internal frictions Q^{-1} were determined for two nondegenerate eigenfrequencies. One depends only on the shear modulus C_{44}. The other, composed of $0.64C_{11}+0.32\ C_{12}+0.04\ C_{44}$, is nearly independent of C_{44}. Granato–Lücke plots of $\ln\epsilon\ Q^{-1}$ vs $\ln\epsilon^{-1}$ showed two regions. The higher strain region showed a slope of approximately 3.5 times that of the lower strain region. A dislocation-model explanation will be offered for higher Q^{-1} at higher ϵ.

10:55

5aPA8. Transducers and measurements at high temperatures using resonant ultrasound spectroscopy. Timothy W. Darling and Albert Migliori (Los Alamos Natl. Lab., Los Alamos, NM 87545)

Resonant ultrasound spectroscopy (RUS) measurements of material properties at high (above 150 °C) temperatures have been directed mainly at materials of geological interest, but a wide variety of other applications exists for high temperature RUS measurements. These include physical properties (phase transitions, elastic moduli) of novel solid state materials and understanding and control of processing parameters for industrial materials such as ceramics and steels. A RUS measurement system has been developed which operates to 600 °C without buffer rods. A description of the materials used and the transducer construction will be presented with some results from measurements on perovskite-type crystal structures and magnetic materials. A discussion of the limits of the present system and possible techniques for bufferless transducers with operating temperatures to 1000 °C will be presented. [Work supported by the U.S. Department of Energy.]

11:10

5aPA9. Resonant ultrasound spectroscopy investigation of a first-order structural phase transition in $LiKSO_4$. F. A. Willis and R. G. Leisure (Dept. of Phys., Colorado State Univ., Ft. Collins, CO 80523)

Lithium potassium sulphate ($LiKSO_4$) exhibits a rich variety of structural phase transitions. Ten different phases have been reported over the temperature range from approximately 4 to 1000 K. The room temperature phase is hexagonal $P6_3$. On cooling, the room temperature phase transforms at approximately 205 K to another phase which is now believed to be trigonal P31c, although the symmetry of this phase has been controversial. Resonant ultrasound spectroscopy has been used to measure the complete set of elastic constants of $LiKSO_4$ over the temperature range of 200 to 300 K. The room temperature elastic constants are accurately described by hexagonal symmetry. At approximately 213 K on cooling there are large abrupt changes of as much as 80% in many elastic constants. The elasticity of this phase cannot be described by hexagonal symmetry, but trigonal symmetry fits the data well. The elastic constants return to the hexagonal values at approximately 243 K on warming. The results will be discussed in terms of Landau theory. [Research supported by NSF under Grant No. DMR-9501550.]

11:25

5aPA10. Ultrasonic spectroscopy techniques used by dolphins to characterize resonating submerged elastic shells. G. C. Gaunaurd, D. Brill, H. Huang (Naval Surface Warfare Ctr., White Oak, Silver Spring, MD 20903-5640), P. W. B. Moore (Naval Command, Control and Ocean Systems Ctr., NRaD, San Diego, CA 92152-6267), and H. C. Strifors (National Defense Res. Establishment, S-17290, Stockholm, Sweden)

The pulsed echoes returned by several submerged cylindrical shells insonified by the peculiar sound pulses ("clicks") emitted by dolphins have been examined. These clicks and echoes were collected in a large database as is done in standard dolphin experiments [*Animal Sonar: Processes and Performance*, edited by P. Nachtigall and P. W. B. Moore (Plenum, New York, 1988)]. The emphasis here is on the processing and physical interpretation of the ultrasonic spectroscopic features in these dolphin-generated echoes. These resonance features actually permit the total characterization of the shells. The spectroscopic "lines" (i.e., resonances with widths) in the frequency signatures, as well as other features in the associated time-domain signatures, provide all the ingredients required to determine the size, shape, thickness, shell elastic material, and internal filler material, in all cases. The time and frequency processing of the echoes is explained in detail; it follows the pattern briefly outlined elsewhere [G. C. Gaunaurd, J. Opt. Eng. **31**, 2253–2261 (1992); G. C. Gaunaurd and H. Huang, J. Inverse Problems Eng. (in press, 1996)], and it will be illustrated with many measured and computationally predicted graphs. These techniques may provide the basic physical explanation of the dolphin's amazing target-ID feats.

11:40

5aPA11. Characterization of inertial confinement fusion targets. Thomas J. Asaki, James K. Hoffer, and John D. Sheliak (MS K764, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Prototype inertial confinement fusion targets for the proposed National Ignition Facility are metallic or plastic spherical shells (2 mm o.d., $\approx$150-mm thickness) with an $\approx$80-mm-thick layer of solid deuterium–tritium (50/50 mixture) deposited on the inner surface. Ignition will occur only if the D–T fuel layer meets strict sphericity and surface roughness criteria (typically $\approx$1 mm). Symmetric layering of solid D–T occurs due to the phenomenon of "beta layering" in which tritium-induced self-heating drives the redistribution of material. In contrast to the optical techniques usually employed, this work discusses methods in which resonant ultrasound spectroscopy (RUS) and related techniques can be used to help determine the uniformity of the fuel layer inside opaque targets. A tetrahedral array of pinducers in a cryogenic apparatus is used to both mount and probe the sample. Preliminary efforts have focused on the characterization of solid spheres and both aluminum and beryllium shells. Sufficiently high Q's (10^3-10^5) necessary for detailed work are readily obtained. Studies involving deuterium-filled shells include the observation of liquid condensation, triple point measurement, and the effects of thermally induced redistribution.

11:55

5aPA12. An application of resonant-ultrasound spectroscopy using a discrete-layer continuum theory. Paul R. Heyliger (Dept. of Civil Eng., Colorado State Univ., Fort Collins, CO 80523), Hassel Ledbetter, and Sudook Kim (NIST, Boulder, CO 80303)

The majority of analysis methods for the computation of natural frequencies required by ultrasonic resonance spectroscopy is based on variational methods of approximation that incorporate basis functions with C^{-1} continuity at every point within the domain. When used for laminated materials of different constitution, the continuous strain field implied by such approximations cannot represent true behavior at an interface between the materials. A discrete-layer theory is developed in this study which introduces piecewise approximations in the layered direction of the solid, allowing for a more accurate portrayal of the strain field. The vibrational modes can be grouped into four groups rather than the eight that are typical for an orthorhombic material. The model is used to study a seven-layer laminate composed of layers of aluminum and aramid/epoxy. Two geometries are considered: a plate and a parallelepiped. The frequencies predicted by the discrete-layer theory are compared with measurements and frequencies found using a more conventional Ritz method with the effective properties of the material. The discrete-layer frequencies tend to be slightly larger than those computed using the Ritz method for the plate and slightly lower for the parallelepiped. Implications for the use of this theory are discussed.

Session 5aPP

Psychological and Physiological Acoustics: Cochlear Mechanics

Peter Dallos, Cochair
Francis Searle Building, Northwestern University, 2299 Sheridan Road, Evanston, Illinois 60208

Glenis R. Long, Cochair
Department of Audiology and Speech Science, Purdue University, West Lafayette, Indiana 47907

Chair's Introduction—8:00

Invited Papers

8:05

5aPP1. A review of active and passive basilar membrane cochlear mechanics. Jont Allen (Acoust. Res. Dept., AT&T Bell Labs., Murray Hill, NJ 07974)

Since David Kemp first proposed the active model in 1979, the role of the outer hair cell (OHC) in basilar membrane (BM) mechanics has been hotly debated. In the "active" model view, the OHC modulates the motion of the BM traveling wave on a cycle-by-cycle basis, leading to a negative BM resistance and a traveling-wave power gain. Nonlinearity is introduced by assuming that the negative BM resistance depends on the signal level. In the "passive" model view, the OHC controls the stiffness of the basilar membrane, leading to a level-dependent (nonlinear) relative impedance between the BM and tectorial membrane and a nonlinear basilar membrane and transduction response. Both models seem to be able to achieve the important nonlinear variations in responses seen in the BM, OHC, IHC, and neural response, but with differing assumptions and degrees of physical reality. It is now clear that the OHC nonlinearly compresses both the dynamic range of basilar membrane motion (Rhode, 1971; Ruggero, 1990) and the neural response (Yates, 1989), extending the otherwise limited dynamic range of the IHC response. This role of the OHC may be quantified using psychoacoustic masking patterns, two-tone suppression, loudness growth and recruitment, and OAEs as objective measures.

8:35

5aPP2. Cochlear nonlinearity: Implications for auditory signal processing and perception. Hendrikus Duifhuis (Biophys. Dept. & BCN (Grad. School for Behavioral and Cognit. Neurosci.), RUG, Nijenborgh 4, 9747 AG Groningen, The Netherlands)

Conscious perception of auditory nonlinearity in combination tones, the low-frequency intermodulation products of two or more tones, goes at least back to Tartini (1692–1770). Renewed interest around 1970 [e.g., J. L. Goldstein, J. Acoust. Soc. Am. **41**, 676–689 (1967); J. L. Hall, J. Acoust. Soc. Am. **56**, 1818–1828 (1974); G. F. Smoorenburg, J. Acoust. Soc. Am. **52**, 615–632 (1972)] led to the conclusion that the nonlinearity originates in the (intact) cochlea. At that time, nobody paid special attention to the point that acoustical effects within the cochlea should be reflected at the aural entrance. Today, that issue (DPOAE) is actively explored. The link with auditory perception receives lesser emphasis, hopefully *not* because a proper psychoacoustic measurement requires more than reading a stimulus parameter (cf. *Psychoacoustics and David M. Green*). Next, two-tone suppression became an issue where psychological and physiological nonlinear acoustics met. The phenomena are intertwined, and probably inseparable from the point of view of underlying biophysical mechanism. Finally, obviously the auditory nonlinearity is compressive. The negligible impact of this result on development of a biophysical basis for the auditory dynamic range and the decibel scale remains puzzling.

9:05

5aPP3. Otoacoustic emissions as tools to probe cochlear function. Glenis R. Long (Dept. of Audiol. and Speech Sci., Purdue Univ., West Lafayette, IN 47907), Carrick L. Talmadge, and Arnold Tubis (Purdue Univ., West Lafayette, IN 47907)

Otoacoustic emissions (acoustic signals originating in the cochlea) can be noninvasively evaluated by placing a sensitive microphone in the ear canal and extracting the emissions from the noise by signal analysis. Since first reported (Kemp, 1978), otoacoustic emissions have been developed into a very useful experimental tool for probing cochlear function. A research overview will give special attention to the existence of a strong periodicity in the frequency domain in all human otoacoustic emissions. This periodicity will be related to maxima and minima seen in hearing threshold measurements. Discussion will focus on the hypothesis that the basis for the pseudoperiodic nature of the emissions is the coherent scattering from cochlear irregularities within the tall and broad peak of the traveling wave [cf. Shera and Zweig, J. Acoust. Soc. Am. **98**, 2018–2047 (1995)]. This scattering gives rise to a reflection of the cochlear wave towards the base, whose effects are detectable in the ear canal signal. The pseudoperiodic properties of emissions are governed by the nearly linear frequency dependence of the phase of the cochlear traveling wave ratio, which is mainly determined by the complex wavelength of the traveling wave in the peak region. [Work supported by NIDCD.]

5aPP4. Mechanical responses of the basilar membrane. Alfred L. Nuttall (Kresge Hear. Res. Inst., Univ. of Michigan, 1301 E. Ann St., Ann Arbor, MI 48109-0506)

The traveling-wave activity that Georg von Békésy studied in the cochlea early in this century has been of intense interest to physiologists and mathematical modelers over the last 30 years. This interest was nurtured by the discovery that the mechanical responses of the basilar membrane are nonlinear at relatively low sound levels and that the nonlinearity is dependent on biological processes [W. S. Rhode, J. Acoust. Soc. Am. Suppl. 1 **49**, S1218 (1971)]. This review will briefly summarize the major findings of the early years as background to a presentation of several issues that are the focus of contemporary work. The steady-state and transient velocity responses of the basilar membrane characterize the capacity of the biological amplification mechanism in the organ of Corti and indicate that a "gain" of 40–60 dB is provided by the outer hair cells. Considerable response distortion accompanies this gain, and wave propagation of certain distortion products occurs. Electrical stimulation of the outer hair cells provides evidence of their mechanical role in the amplification mechanism by demonstration of the frequency range and displacement capacity of the cells. Efferent innervation of the organ of Corti is found to modulate the mechanical activity of the system.

10:05–10:20 Discussant

George Zweig

Los Alamos Natl. Lab., MS B276, T-Dot, P.O. Box 1663, Los Alamos, NM 87545

10:20–10:30

Open Discussion

Contributed Papers

10:30

5aPP5. Inner hair cell and organ of Corti responses to very low frequency tones. M. A. Cheatham and P. Dallos (Hugh Knowles Ctr., 2-240 Frances Searle Bldg., 2299 N. Campus Dr., Northwestern Univ., Evanston, IL 60208)

The timing of excitation observed in the auditory nerve exhibits a strong dependence on best frequency (BF). Fibers innervating the base of the cochlea respond to near threshold, low-frequency inputs approximately in phase with basilar membrane velocity to scala tympani while fibers innervating apical regions respond in phase with basilar membrane velocity to scala vestibuli [M. A. Ruggero and N. C. Rich, J. Neurophysiol. **58**, 379–403 (1987)]. Although the latter is consistent with the velocity dependence of inner hair cells (IHC) and with the classical view of hair cell stimulation, the response phase of single units in the base of the cochlea has been difficult to explain. Consequently, IHC recordings from the second turn of the guinea pig cochlea (BF = 4000 Hz) were used to determine phases of depolarization relative to basilar membrane displacement. To better estimate synaptic drive, the organ of Corti response was subtracted from that recorded in the IHC since the voltage gradient that induces transmitter release can be influenced by extracellular responses that reflect receptor currents generated in nearby outer hair cells. Results indicate that IHC depolarization occurs near basilar membrane velocity to scala tympani consistent with single unit excitation in the base of the cochlea. [Work supported by Grant No. 5R01DC00089, National Institute on Deafness and Other Communication Disorders.]

10:45

5aPP6. Impact noise exposure in chinchilla: Effects of frequency, level, and interstimulus interval. Rickie R. Davis and William J. Murphy (Bioacoust. and Occup. Vib. Sect., Natl. Inst. for Occup. Safety and Health, MS C-27, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

The effect of impact noise on hearing threshold was investigated in a parametric study which manipulated frequency, level, and interstimulus interval (ISI). The chinchilla pinna passively amplifies acoustic stimuli between 2 and 6 kHz suggesting greater effects in that frequency range. Exponentially decaying pulses (t = 40 ms) with carrier frequencies of 1, 3, and 8 kHz were presented at peak pressure levels of 114, 117, 120, and 129 dB SPL and ISI of 1, 10, and 100 s. The chinchillas were exposed for 8 h, with one animal per condition. Hearing function was assessed by measuring auditory-evoked potentials from tonebursts at 0.5, 1, 2, 4, and 8 kHz from implanted electrodes. The data indicate a small (< 10 dB) permanent threshold shift (PTS) in the 10- and 100-s ISI conditions. For the 1-s ISI conditions, the animals exposed at 129 dB experienced more than 25 dB PTS at high frequencies. Animals exposed at 120 dB exhibited large temporary threshold shifts (TTS) while the animals exposed at lower levels had smaller TTS and minimal PTS. Stimulus level and ISI had significant effects on the PTS while frequency content was not significant.

11:00

5aPP7. Neural representation and psychophysical discrimination of vowel formants. Bradford J. May (Dept. of Otolaryngol.–HNS, Johns Hopkins Univ., Baltimore, MD 21205) and Robert D. Hienz (Johns Hopkins Univ., Baltimore, MD 21205)

Auditory-nerve fiber responses were modeled to estimate neural representations of frequency changes in the second formant (ΔF_2) of the steady-state vowel /eh. Simulations were performed at vowel levels of 33–84 dB, both in quiet and in continuous noise at a constant 3-dB signal-to-noise ratio (S/N). Signal detection analysis of model outputs suggested that formant changes at higher stimulus levels and in background noise were better represented by rate responses of fibers with low spontaneous rates (SR). Psychophysical tests were then performed in cats to measure behavioral thresholds for the detection of formant frequency changes under stimulus conditions similar to those of the neural simulations. Behavioral ΔF_2's were obtained at vowel levels of 10–70 dB, in quiet and in continuous background noise at S/Ns of 3, 13, and 23 dB. ΔF_2 decreased with increasing vowel level and increased with decreasing S/N. These trends in psychophysical performance paralleled changes in the quality of vowel representations, particularly those based on discharge rates of low SR auditory-nerve fibers. [Work supported by NIDCD Grant Nos. 5R01DC01388-04 and 2R01DC00109-22.]

11:15

5aPP8. Perception of amplitude fluctuation in the goldfish. M. Chronopoulos, R. Fay, R. Dye (Parmly Hear. Inst. and Dept. of Psych., Loyola Univ.–Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626), S. Sheft, and W. Shofner (Parmly Hear. Inst., Chicago, IL 60626)

This experiment investigates the extent to which the goldfish's response to amplitude-modulated and phase-manipulated sounds can be predicted by the power of envelope fluctuation. Sixteen common goldfish, *Carassius auratus*, were classically conditioned to suppress respiration to a 350-Hz carrier 100% sinusoidally amplitude modulated at 30 Hz. In a subsequently generalization test session, respiratory suppression was measured in the presence of novel signals having either sidebands attenuated or carrier phases shifted up to 90 °. Respiratory suppression declined with both increases in sideband attenuation and increases in phase shift of the carrier frequency. Respiratory responses to both sideband attenuated and phase-shifted signals declined along a single, monotonic function of envelope power as defined by the normalized fourth moment [W. Hartmann and J. Pumplin, J. Acoust. Soc. Am. **90**, 1986–1999 (1991)]. The goldfish's performance in this experiment is comparable to the performance of human listeners in experiments on roughness scaling [C. Mathes and L. Miller, J. Acoust. Soc. Am. **19**, 780–797 (1947)]. [Work supported by a NIH, NIDCD Program Project Grant to the Parmly Hearing Institute.]

11:30

5aPP9. Perception of musical pitch with electrical stimulation of the cochlea. Hugh J. McDermott and Colette M. McKay (Dept. of Otolaryngol., Univ. of Melbourne, Parkville, 3052 Australia)

For cochlear implant users to obtain enjoyment from music, it is essential that they be able to perceive melodic pitch. Previous studies have indicated that the "pitch" or timbre of pulsatile electrical stimuli varies with the repetition rate or intracochlear site of delivery, but generally pitch has been defined loosely as a tonal quality that can be ranked in terms of "sharpness." In a recent series of experiments, a user of the Nucleus 22-electrode implant was asked to judge or adjust the musical interval between pairs of stimuli. The electrical parameters investigated included the frequency of sinusoidally amplitude-modulated pulse trains, as well as pulse rate and active electrode position. The subject was able to judge the pitches using only the interval names ("fifth" "octave," etc.), and without specific training, the guidance of familiar melodies, or rhythm cues. The results showed that rate or modulation frequency could convey musical pitch over a range of approximately two octaves, and followed a relationship comparable with that for acoustic stimuli and normal hearing. The pitch related to electrode position could also be labeled in musical terms. Interestingly, when both place and rate varied together, the pitch associated with electrode position was generally dominant.

11:45

5aPP10. The effect on pitch and loudness of major interpulse intervals within modulated current pulse trains in cochlear implantees. Colette M. McKay and Hugh J. McDermott (Dept. of Otolaryngol., Univ. of Melbourne, Parkville, 3052 Australia)

Two experiments were completed with five subjects who have been implanted with the Mini System 22 cochlear implant. In the first experiment, 1000-Hz pulse trains were modulated so that there were two high- and eight low-current pulses in each 10-ms period. Five stimuli were constructed with differing time intervals between the two high-current pulses, and a further six stimuli were unmodulated pulse trains with rates between 100 and 250 Hz. All stimuli were loudness balanced and presented to the subjects in a single-interval pitch ranking task. The unmodulated stimuli showed a monotonic increase in pitch with rate. The pitch of the modulated stimuli was mostly determined by the longer of the two intervals between the high currents. When the high currents were very close, the modulation period was also important in determining the pitch. The second experiment used constant-current pulse trains with two pulses in every 20 ms. The effect on loudness of the interval between the two pulses was measured using loudness balancing. For interpulse intervals less than 4 ms the loudness increased as the interval decreased. This result implies that a facilitatory effect dominates over any refractory effect on loudness at short interpulse intervals.

12:00

5aPP11. Effects of meclizine in young adults, measured with otoacoustic emissions, REPs/ABR, qEEG, and a computerized test of eye–hand coordination. Judith L. Lauter, Suzanne B. Wood, Onita Lynch, and Kaye Agnew (Dept. of Commun. Sci. and Disord., P. O. Box 26901, 825 N.E. 14th St., Univ. of Oklahoma Health Sciences Ctr., Oklahoma City, OK 73190)

Twelve neurologically normal young adults were tested before and after administration of meclizine, an over-the-counter medication for motion sickness. The battery consisted of four components: (1) repeated-measures distortion-product otoacoustic emissions (DPOAEs); (2) the repeated-evoked-potentials version of the auditory brain-stem response (REPs/ABR); (3) quantitative electroencephalography (qEEG) measured over left- and right-side auditory cortex; and (4) a computer-based eye–hand coordination task. The battery required approximately 1.5 h to complete. Each subject was tested with the battery in each of eight longitudinal sessions: three times on a control day (no medication); the same times on a second day one week later (medication at mid-day); and 24- and 48-h check-up sessions following the medication day. Results show dramatic changes in all battery components, with details suggesting the site(s) of action of this type of antihistamine. The "auditory-system cross sections" yielded by this battery make it possible to observe effects from periphery to cortex, including evidence linking otoacoustic emissions with central auditory physiology. Implications range from cautions regarding the acute effects of antihistamines, to physiological support for using such medications to improve performance in learning-disabled children.

Session 5aSA

Structural Acoustics and Vibration: Acoustics of Coupled Structures I

J. Adin Mann III, Chair
Aerospace Engineering and Engineering Mechanics, Iowa State University, Ames, Iowa 50011

Chair's Introduction—8:00

Invited Papers

8:05

5aSA1. Joint effects in the mid-frequency vibration of connected plates. T. Igusa (Dept. of Civil Eng., Northwestern Univ., Evanston, IL 60208)

Structure-borne noise in coupled systems has been shown to be highly sensitive to joint properties. In this presentation, the effects of joint behavior is examined for coupled plates joined in the canonical L-shaped configuration. The effects of the variation of the joint properties with respect to the coordinate axis along the two-plate junction is of interest. Mid-frequency loads are used, where the modal density is not sufficiently high for standard statistical energy analysis. The problem is analyzed using several methods. A mobility approach [J. M. Cuschieri, J. Acoust. Soc. Am. **87**, 1159–1165 (1990)] is used as a benchmark, which, for the thin-plate model, is exact except for mode truncation. The problem is then examined in the perspective of the following two approximate methods: a spatial and temporal averaging method using energy and intensity as the primary field variables [O. M. Bouthier and R. J. Bernhard, AIAA J. **30**, 616–623 (1992)] and an asymptotic method, based on Skudrzyk's Mean Value Theory [T. Igusa and Y. Tang, AIAA J. **30**, 2520–2525 (1992)]. It is found that the effects of the variations in the joint properties can be interpreted using wave vector decompositions.

8:35

5aSA2. Structural acoustic energy finite element method. Fernando Bitsie and Robert J. Bernhard (1077 Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

A new numerical method, referred to as the energy finite element method, had been previously developed to predict the high-frequency response of built-up structural acoustic systems consisting of subsystems such as rods, beams, plates, and acoustic spaces. The methodology for prediction of behavior in the subsystems is based on a diffuse energy field approximation. Subsystems are coupled together using net energy flow and energy superposition principles. In this paper, the structural acoustic coupling relationship for energy flow analysis of coupled plates and acoustical spaces is shown. The coupling approach chosen for structural acoustic radiation uses plate radiation efficiency. The approach has been implemented into an energy finite element model. The energy finite element predictions will be compared to measured results for an acoustical enclosure with one flexible wall.

9:05

5aSA3. Fluid-loaded plate coupled to a distributed inhomogeneity—Review. Joseph. M. Cuschieri (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431) and David Feit (David Taylor Res. Ctr., Bethesda, MD 20084)

The problem of scattering from a fluid-loaded plate structure coupled to a distributed inhomogeneity is considered. Results have been presented which include the plate surface velocity Green's function, the far-field and near-field scattered pressure, and the acoustic and structural intensity. The distributed inhomogeneities considered varied both in type (mass or stiffness) and shape. In all instances, the perturbation due to the inhomogeneity, relative to the elastic characteristics of the plate, is small. In the case of shape variations, the influence of "smoothness"at the edges of the inhomogeneity, and of oscillations within the inhomogeneity distribution were also considered. In this presentation, the results that have been generated thus far, will be reviewed and general conclusions drawn from these results. It is shown that depending on the frequency range of interest, the results can be significantly influenced by the shape and type of inhomogeneity. Stiffness inhomogeneities, in general, are less significant compared to mass inhomogeneities. However, in relative terms, internal variations in the inhomogeneity shape are more significant for stiffness inhomogeneities than for mass inhomogeneities. Alternate forms of presenting the results will be explored. [Work sponsored by ONR.]

9:35

5aSA4. The vibro-acoustic response of coupled structures using hybrid analytic-numeric methods. Karl Grosh and Peter J. Halliday (Dept. of Mech. Eng. and Appl. Mechanics, Univ. of Michigan, Ann Arbor, MI 48109-2125)

A hybrid analytic-numeric formulation for the time harmonic structural acoustics problem is presented. The variational framework for the seamless inclusion of analytic solutions into the Galerkin finite element formulation for a fluid-loaded structure and results of the application of this method are given. One goal of this formulation is to increase overall efficiency by eliminating regions of the structure from the computational domain via an analytic representation of the response. In this way, the total degrees of freedom present in the problem will be reduced. This reduction in the degrees of freedom enables higher frequency, more complex problems to be solved by decreasing memory requirements and compute time. The analysis of complex structures, comprised of structural members coupled at joints, is facilitated by this approach. Using a discretization of the equations of elasticity for the joint, its response may be represented to a desired level of accuracy. The elasticity representation is coupled to the reduced plate or shell theory producing an efficient model of the complete structure. The effect of the joint and the level of modeling detail required for the desired fidelity of prediction for the vibration and acoustic response of the system are studied.

10:05–10:20 Break

Contributed Papers

10:20

5aSA5. Multiple-reference adaptive active noise control in enclosures. Zane M. Rhea (Grad. Prog. in Acoust. and Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804), Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT 24360), and Courtney B. Burroughs (Penn State Univ., State College, PA 16804)

Noise control of multiple noise sources in 3-D enclosures remains a concern in many applications. This is of particular interest in vehicles, where multiple primary sources contribute to the sound field inside the enclosure. When utilizing active noise control (ANC) systems in these applications, multiple reference inputs are required. A comparison of active noise control with multiple and single reference inputs of multiple noise sources is presented in a case format. Since most sources are structural in origin, independent harmonic vibration of two of the rectangular enclosure walls are used as the primary noise sources. Although a multiple reference input ANC system is much more successful at controlling multiple noise sources, care must be taken in choosing the reference inputs. If the reference inputs are not independent the ANC becomes ineffective. Also, a case study of sound-pressure reductions obtained for on and off resonance of both the enclosure and the enclosure walls is presented. [Work supported by the Applied Research Laboratory.]

10:35

5aSA6. Sound transmission through an aeroelastic plate into an acoustic cavity. Robert L. Clark and Kenneth D. Frampton (Dept. of Mech. Eng. and Material Sci., Duke Univ., Durham, NC 27708-0300)

The transmission of turbulent boundary layer (TBL) pressures into acoustic enclosures is very important to the aerospace industry. This topic is particularly important in aircraft interior noise investigations. While many studies have been published concerning TBL transmission through elastic plates into acoustic enclosures, few of these studies have considered the dynamic effects of aerodynamic flow over the plate (i.e., aeroelasticity). This presentation investigates the modeling of an elastic plate subject to full potential flow and TBL loading on one side and coupled to a reverberant acoustic enclosure on the other side. The elastic plate and acoustic enclosure are modeled through a Rayleigh–Ritz approach. The effects of the external fluid flow are modeled through a singular valued decomposition technique which performs a system identification on an approximate numerical solution to the full potential flow equations. This aerodynamic model is then coupled to the elastic plate/cavity model to form the complete system. The primary focus of this investigation is the effects of the external flow on the sound transmission through the elastic plate into the cavity. It is demonstrated that external flow can significantly affect the system dynamics and, hence, the sound transmission.

10:50

5aSA7. Stability of clamped rectangular plates in uniform subsonic flow. Jinshuo Zhu (Perstorp Components, 47785 W. Anchor Ct., Plymouth, MI 48170) and Sean F. Wu (Wayne State Univ., Detroit, MI 48202)

This paper depicts the stability charts of rectangular plates clamped to an infinite, rigid baffle in uniform subsonic flow. The correlations among the critical flow speeds and the plate aspect ratio, plate thickness/length ratio, and plate/fluid density ratio are exhibited. Results show that when the flow speed exceeds a critical value, the plate may vibrate around an equilibrium position other than its undeformed one. When the flow speed exceeds all the critical values, the plate may be locally unstable at all equilibrium positions. In particular, it may jump from one equilibrium position to another in a random fashion. These local instabilities are controlled by structural nonlinearities. Without the inclusion of structural nonlinearities, the plate may have only one equilibrium position, namely, its undeformed one. The amplitude of plate vibration would then grow unboundedly when the flow speed exceeds the critical value, known as absolute instability. With the inclusion of structural nonlinearities, the plate may have more than one equilibrium position when the flow speed exceeds the critical values. Under this condition, plate vibration may seem chaotic, the overall amplitude of flexural vibration is nevertheless bounded.

11:05

5aSA8. Enhancement of the transmission loss of panel structures through the application of segmented, resonant foam attachments. J. Stuart Bolton and Yeon June Kang (1077 Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-1077)

In noise control foam both the bulk solid and fluid phases participate significantly in the wave propagation process. Here it is suggested that resonances of the foam's solid phase can be tuned to produce significant narrow-band absorption superimposed upon the broadband attenuation characteristic typically offered by the fluid phase of an elastic porous material. Two approaches have been considered. In the first, aluminum masses are applied to small pieces of foam (say, 25 mm square and 6 mm deep) that are then attached to a panel. Each piece of foam then comprises a small resonator whose natural frequency is essentially determined by the bulk stiffness of the solid phase of the foam together with the added mass. It has been found that the application of an array of foam resonators causes the transmission loss of the base panel to be increased over approximately an octave. In a second approach, a continuous foam lining is segmented, and a piece of perforated metal is applied to each segment. Again, a narrow-band increase in transmission loss is observed, in this case combined with the broadband transmission loss increase normally produced by a layer of porous material.

11:20

5aSA9. Discontinuous constrained layer damping treatments applied to a vibrating free-free beam. Samir Uppal, Alison B. Flatau (Dept. of Aerospace Eng. and Eng. Mechanics, Iowa State Univ., Ames, IA 50011), and Theodore B. Bailey (Iowa State Univ., Ames, IA 50011)

The vibration response to application of discontinuous constrained layer damping (CLD) patches to varied portions of a free-free beam was studied. Since neither closed-form solutions nor finite element methods can be readily implemented for CLD analysis with partial coverage of a structure under broadband excitation [C. T. Sun and Y. P. Lu, *Vibration Damping of Structural Elements* (Prentice-Hall, Englewood Cliffs, NJ, 1995), pp. 318–362], an empirical approach was taken. The test object was a 26.5- × 1- × 1/8-in. steel beam, and was subjected to a chirp. The aim was to quantify the relative effectiveness of varied lengths and positions of CLD patches in reducing the response of the first five resonant modes of the beam. Assessment of the potential for strain energy dissipation was made based on the net displacement of the beam (appropriately phased mode summation), with relative phase of the summed modes specific to the

input force signal. Attempts were made to correlate the effect of CLD on each mode to the time-averaged percentage of strain energy for that mode under the treatment patch, coupled with the percentage of the beam's strain energy under the patch. [Work supported by NSF.]

11:35

5aSA10. Spectral elements for acoustic wave propagation through thin-walled complex structures. Brian A. Bilodeau and James F. Doyle (School of Aeronaut. and Astronaut., Purdue Univ., West Lafayette, IN 47907)

Noise generation is an aspect of structural impacts not receiving much attention. Reasons for this include the fact that the problem requires analyzing the solid/fluid interaction as well as the need to model extended regions. Spectral elements are developed for analyzing acoustic wave propagation through thin-walled structures. Both flat- and curved-plate elements, which incorporate the effect of fluid loading on the structure, are developed. These elements exactly model the responses over large domains and through their generalized nodes may be conveniently joined to model complex structures composed of many segments and faces. Results for the impact of flat and curved panel systems are presented. In addition, the utility of the spectral method for computing frequency response functions directly is demonstrated.

11:50

5aSA11. Spectral integral approach to radiation from fluid-filled boreholes. Rama Rao V. N., Henrik Schmidt, and J. Kim Vandiver (MIT, Dept. of Ocean Eng., 77 Massachusetts Ave., Rm. 5-007, Cambridge, MA 02139)

A hybrid model for radiation from a borehole while drilling has been developed. A numerical model for propagation in a borehole including a drill pipe has been combined with the OASES frequency–wave-number model for propagation in layered earth media, using the concept of effective sources. Equivalent line source arrays that produce the same far-field radiation as the borehole are first computed. These are then introduced into the layered earth model and the resultant seismic wave field is computed. The borehole modes are excited by a displacement source at the bottom, simulating bit motion while drilling. In contrast to earlier work the present model includes the drill pipe in the simulation, allowing for three propagating modes at low frequencies. The model has been applied to analysis of the role of the formation composition in the radiation from the tube modes. The fundamental differences between the radiation into soft and hard formations are discussed and demonstrated, including the formation of Mach cones along the borehole in soft formations.

FRIDAY MORNING, 17 MAY 1996 MT. RAINIER AND MT. MCKINLEY, 8:30 TO 11:30 A.M.

Session 5aSC

Speech Communication: Speech Perception, Word Recognition and Talker Characteristics (Poster Session)

Keith A. Johnson, Chair

Department of Linguistics, The Ohio State University, 1712 Neil Avenue, Columbus, Ohio 43210-1298

Contributed Papers

All posters will be on display from 8:30 to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 to 11:30 a.m.

5aSC1. Voice effects in implicit memory tasks. Julie M. Brown, Carol A. Fowler, and Jay G. Rueckl (Dept. of Psych., 406 Babbidge Rd., Box U-20, Univ. of Connecticut, Storrs, CT 06269 and Haskins Labs., New Haven, CT 06511)

Spoken words are easier to identify if they have been heard recently. This phenomenon, known as repetition priming, can be used to investigate the processes underlying word recognition. Using an implicit memory paradigm, this study looked at the effect of changing the voice of the speaker on repetition priming. Voice effects occur if repetition priming is reduced when a spoken word has been heard in different voices at study and test. Voice effects have been found in implicit memory tasks, such as word identification and word-stem completion [B. A. Church and D. L. Schacter, JEP:LMC **20**, 521–533 (1994); S. D. Goldinger, doctoral dissertation, 1992]. This study investigated if these results could be generalized to other implicit memory tasks—specifically, lexical decision, naming, and a new task, auditory fragment completion. No voice effects were found with any of these three tasks. These results are problematic for current accounts of word recognition and imply that as yet unidentified factors control the occurrence of voice effects.

5aSC2. Possible word boundary constraints on multiple activation of form-based representations of spoken words. Paul A. Luce, Rochelle S. Newman, and Emily A. Lyons (Dept. of Psych., Park Hall, SUNY, Buffalo, NY 14260)

Most current theories of spoken word perception (e.g., Cohort theory, TRACE, Shortlist, Neighborhood Activation Model) propose that multiple form-based representations of words are activated in memory during recognition. Few of these theories, however, specify the precise nature of multiple activation of words in fluent speech or the constraints on activation imposed by information within the speech signal. The degree to which word boundary information in two-word sequences may limit lexical hypotheses during recognition was examined. Using a cross-modal priming technique, spoken two-word utterances, such as NOTE-RAIL, were presented and the activation of possible lexical items spanning the word boundary (e.g., TRAIL) was investigated. The implications of these results for current theories of spoken word recognition will be discussed. [Work supported by NIDCD.]

5a FRI. AM

5aSC3. The segmental representation of words as revealed by priming in a lexical decision and naming task. Nancy J. Palmer and James R. Sawusch (Park Hall, SUNY, Box 604110, Buffalo, NY 14260-4110)

In previous studies, a priming task was used to explore the nature of the segmental representation of words. The phonetic overlap between primes and targets was varied. When phonemes occurred in the same syllable position in prime and target, responses to the target were faster than when prime and target shared no common phonemes. In contrast, when prime and target shared phonemes but the phoneme positions were different, no facilitation of target responses was found. New studies have further examined the segmental representation of speech using a new set of stimuli with both naming and lexical decision tasks. While previous prime–target pairs have been either both nonwords or words, the new studies include word–nonword and nonword–word trials. Additionally, the vowels in CVC prime and target were varied to examine the influence of the vowel similarity on priming when prime and target have consonants in common (e.g., prime "cat," target "kit"). Results will be discussed in terms of their implications for the nature of the abstract, segmental representation that underlies word recognition and lexical access. [Work supported by NIDCD Grant No. R01 DC00219 to SUNY at Buffalo.]

5aSC4. Building lexical neighborhoods. Michael S. Vitevitch and Paul A. Luce (Dept. of Psych., SUNY, Buffalo, NY 14260-4110)

The Neighborhood Activation Model (NAM) states that the speed and accuracy of spoken word recognition are a function of the number and nature of neighbors—or similar sounding words—activated in memory by stimulus input. In particular, the model states that spoken word recognition is a function of (1) target word frequency, (2) similarity neighborhood density, and (3) similarity neighborhood frequency. In a pretest–training –posttest design, these three factors were manipulated using nonword target words and nonword neighbors to test directly predictions of NAM. The results were only partially consistent with the model. As predicted, processing times to high-frequency targets increased as a function of neighborhood density. However, processing times for low-frequency targets actually decreased with increased neighborhood density. Furthermore, no significant effects of neighborhood frequency were obtained. The implications of these findings for NAM and other models of spoken word recognition are discussed.

5aSC5. The neighborhood characteristics of malapropisms. Michael S. Vitevitch (Dept. of Psych., SUNY, Buffalo, NY 14260-4110)

This study examined the neighborhood characteristics (frequency, density, and neighborhood frequency) of 138 malapropisms (whole word errors). A qualitative analysis suggests that words that are more often involved in malapropisms have low neighborhood frequencies. Additionally, these words tend to be either high-frequency target words in dense neighborhoods, or low-frequency target words in sparse neighborhoods. Finally, malapropisms tended to occur such that the word that was substituted for the intended word was more frequent than the intended word. The implications of these findings for models of lexical representation and processing are discussed.

5aSC6. The effect of talker rate and amplitude variation on memory representation of spoken words. Ann R. Bradlow (Speech Res. Lab., Dept. of Psych., Indiana Univ., Bloomington, IN 47405) and Lynne C. Nygaard (Emory Univ., Atlanta, GA 30322)

This study investigated the encoding of spoken word attributes in a task of continuous recognition memory. In experiment 1, subjects were presented with a list of spoken words, and for each word they judged whether the word was "old" (had occurred previously in the list) or "new." Results showed that subjects were more accurate at recognizing a word as "old" if it was repeated in the same voice (condition 1), and at the same rate of speech (condition 2); however, no recognition advantage was found for words repeated at the same amplitude (condition 3). In experiment 2, subjects gave an additional explicit judgment as to whether "old" words were repeated in the same voice (condition 1), rate (condition 2), or amplitude (condition 3). Subjects again showed a recognition advantage for words repeated in the same voice and rate, but no advantage occurred in the amplitude condition. However, subjects in all three conditions were significantly above chance accuracy in recognizing whether an "old" word was repeated in the same voice, rate or amplitude. These data suggest that, even though voice, rate, and amplitude information may be encoded differently, information along all three dimensions is retained in memory. [Work supported by NIH.]

5aSC7. Implicit memory for silent-center syllables: Effects of talker, token, and delay. Susan L. Hura, Lauren E. Chasey, Joshua S. Kopf, and Emily C. Muelhausen (Purdue Univ., Dept. Audiol. and Speech Sciences, 1353 Heavilon Hall, West Lafayette, IN 47907-1353)

Listeners' ability to reconstruct missing vowel information in silent-center (SC) syllables has been viewed as evidence that vowels are dynamically specified by the information in syllable transitions. In this study implicit memory techniques are used to explore the representations formed when listeners perceive SC syllables. A set of CVC words spoken by several talkers was used to create SC and full syllable stimuli. A subset of SC stimuli was presented to listeners for identification. After a variable delay listeners were tested on the complete set of full syllables, some of whose SC versions were presented in the initial phase. Other full syllables overlap with the SC set in terms of talker or token. Performance on full syllables with differing degrees of overlap between SC and full stimuli is compared across a range of delays. Facilitation on full syllable stimuli is viewed as evidence that representations of individual SC stimuli are stored in memory and accessed in later perception of full syllables.

5aSC8. Perceiving the sex and identity of a sine-wave talker. Jennifer M. Fellowes, Robert E. Remez (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027-6598), and Philip E. Rubin (Haskins Labs., New Haven, CT 06511)

Listeners can readily perceive both the linguistic message and the identity of the talker from an utterance that has been replicated with a few time-varying sinusoids. One puzzling outcome of studies of the identification of sine-wave talkers is that erroneous identifications did not cluster by sex. Males were often mistaken for females. and vice versa. A series of new experiments used frequency transpositions of sinusoidal replicas of speech to determine the acoustic attributes responsible for the identification of a talker's sex and a talker's identity. The central spectral tendency of a sinusoidal sentence was found to affect the perception of the sex of the talker; a sine-wave pattern derived from the formant frequencies of an individual talker seemed male, female, or neither when transposed to match the male, female, or overall average formant frequencies in our talker set, respectively. However, performance on a test of individual recognition was very good under acoustic conditions that reduced sex determination to chance levels. This suggests that recognizing a sine-wave talker does not depend on prior identification of sex, and begins to explain why listeners are likely to confuse sine-wave talkers without regard for the talker's sex. [Work supported by NIDCD.]

5aSC9. Talker variability effects on spoken word recall: Encoding or rehearsal? Meral Topcu, John W. Mullennix, Tina Bakalis, Maria Stauch, and Tina Tracevski (Dept. of Psych., Wayne State Univ., Detroit, MI 48202)

The effects of talker variability were examined for spoken word recall. In this study, a serial word recall paradigm was used, with talker variability manipulated using single- and multiple-talker lists and with presentation rate varied using fast and slow conditions. In addition, memory rehearsal processes were examined via a continuous distractor task. Under no-distractor conditions, the results showed that primacy recall performance was better for single-talker lists compared to multiple-talker lists. Under distractor conditions, when presentation rate was fast, this difference remained. However, when presentation rate was slow, there was little difference in recall for single-talker versus multiple-talker lists. These results will be discussed in terms of the roles of rehearsal and encoding processes in the transfer of voice information to episodic memory for spoken words. [Work supported by NIH Grant No. DCO1667-03.]

5aSC10. Learning voices. Lynne C. Nygaard (Dept. of Psych., Emory Univ., Atlanta, GA 30322) and David B. Pisoni (Indiana Univ., Bloomington, IN 47405)

The present experiment was designed to investigate the perceptual learning of talker's voice. Listeners were trained over a 9-day period to become familiar with a set of ten talkers (five male and five female). During each day of training, listeners were asked to identify talkers' voices from isolated words and to associate each voice with a common name. On the tenth day of testing, listeners were given a generalization test which consisted of a set of novel words produced by the ten talkers they heard in training and by ten unfamiliar talkers (five male and five female). The results showed that over 9 days of training, talker identification performance improved. However, there were large individual differences among listeners in talker learning performance. In addition, individual differences in the identifiability of voices was observed. Male voices were identified significantly better than female voices by both male and females listeners. The results of the generalization task showed that the introduction of unfamiliar voices reduced identification performance for familiar talkers. Listeners were able to judge overall familiarity, but confusions increased in identification of familiar talkers. The results suggest that talker-, listener-, and task-specific factors influence the perceptual learning of talker's voice.

5aSC11. The effects of multiple-talker stimuli on immediate memory span. Helena M. Saldaña and William R. Svec (Dept. of Psych., Indiana Univ., Bloomington, IN 47405)

Previous experiments revealed that changing the voice between study items on an auditory list of letters results in a decrease in memory span [H. M. Saldaña, J. Acoust. Soc. Am. (1995)]. This finding is contrary to Baddeley's articulatory loop model which defines memory span in terms of time or duration of items. According to this model items are stored as memory traces that fade after approximately 2 s unless revived by an articulatory control process [A. D. Baddeley and G. J. Hitch, Psychol. Learn. Motiv. (1974)]. The number of items that can be reactivated within the decay time can be retained indefinitely. The present study replicates and extends the previous memory span findings by utilizing digits instead of letters and by using a stimulus matching methodology for estimating span length. A reconceptualization of the articulatory loop model is discussed which takes into account the encoding of stimulus variability. [Work supported by NIH.]

5aSC12. Perceptual compensation for vowel undershoot may be explained by general perceptual principles. Lori L. Holt, Andrew J. Lotto, and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706)

When presented a series of /dVd/ syllables varying perceptually from /dɛd/ to /dʌd/, listeners identify the vowel sound as /ʌ/ more often than when the vowels are presented alone as steady states. Conversely, /bVb/ series lead to more /ɛ/ identifications. This pattern of responses appears to demonstrate a compensation for assimilatory effects of coarticulation or vowel "undershoot." Experiments will be presented which test the specificity of this perceptual process. Two ten-step (/bVb/ and /dVd/) series of 100-ms /CVC/ syllables varying in $F2$ midpoint frequency (1260–1710 Hz) were synthesized. Nonspeech analogs were created by appending 70-ms sine-wave glides, which followed a similar trajectory to $F2$ in the speech stimuli, to 60-ms steady-state vowels varying in $F2$ frequency. A second set of nonspeech series was created by appending constant-frequency sine-wave tones to the vowel stimuli. The frequency of these tones were set at the $F2$ starting frequencies for the speech stimuli. These nonspeech contexts had effects on vowel identification that were analogous to those for the speech contexts. Lower frequency glides and tones (modeling /bVb/$F2$) led to more /ɛ/ (higher $F2$) identifications with the complementary pattern for higher frequency glides and tones. [Work supported by NIDCD and NSF.]

5aSC13. Identification of vowels based on visual cues within raw complex speech waveforms. Michael A. Stokes (Indiana State Dept. of Health, P. O. Box 241153, Indianapolis, IN 46224)

Testing was performed to demonstrate that subjects can identify vowels using visual displays of raw complex waveforms. Two Midwestern males produced nine vowels [Peterson and Barney, J. Acoust. Soc. Am. **24**, 175–184 (1952)] for two identification trials. In both trials, nine vowels were presented in random order by an experimenter. Subject MS correctly identified five out of nine vowels in both trials. Identification was accomplished by recognizing temporal interactions between F0 and F1 that provide categorical boundaries across the vowel space. The presence or absence of F2 in the range of 2000 Hz also provides a distinguishing characteristic between categorical pairs. When formant values and articulatory data are organized by categories seen within speech waveforms, the vowel space resembles the pairings of the stop consonants /b/–/p/, /d/–/t/, and /g/–/k/. Tongue position (reflected in F1 values) categorizes a vowel as place of articulation categorizes stop consonants. Lip position (reflected in F2 frequency) distinguishes vowels within a category as voicing distinguishes these consonant pairs. The details and potential success of a new model of vowel perception based on these findings will be discussed.

5aSC14. Enhancement of concurrent vowels: Postcursor effects and thresholds. D. Dwayne Paschall (Dept. of Commun. Disord., Texas Tech Univ. Health Sciences Ctr., Box 42073, Lubbock, TX 79409-2073)

Speech communication often takes place in noisy environments, including situations where two or more voices compete for the attention of the listener. Experiments with simultaneous vowel pairs investigated aspects of a priming stimulus that draws attention to a target voice in a mixture of two voices. Previous studies have shown identification accuracy is higher in conditions where the vowel pair was preceded by a precursor with the same F_0 and spectrum envelope as one of the vowels. These findings are consistent with an explanation based on auditory adaptation. This conclusion is reinforced by results from the present experiments which show that when the precursor is placed *after* the concurrent vowel pair (a postcursor), there is little benefit to identification accuracy. Thresholds for identifying a target vowel in the presence of a masker vowel when a precursor is present with the same intensity, F_0, and spectrum envelope as the masker vowel suggests that the phenomenon of enhancement appears consistently only when the enhanced vowel and the precursor are at similar intensities. The present experiments suggest that enhancement is the result of perceptual grouping mechanisms that are supplemented by adaptation processes possibly operating at a level more central than the auditory nerve.

5aSC15. Vowel perception in consonant context. Rachel Thorburn (Linguist. Dept., Univ. of Massachusetts, Amherst, MA 01003)

Macmillan, Goldberg, and Braida [J. Acoust. Soc. Am. **84**, 1262-1280 (1988)] show that when stimuli are presented along a continuum, instances of consonants that are near good exemplars are most memorable for labeling by a listener. However, for vowels (produced in isolation), instances which are near category boundaries are most memorable. Can these results be generalized to vowels produced in context? What is the effect of different consonant contexts on vowel perception? To answer these questions, this study tested high vowels along a front–back continuum produced in two CVC contexts (labial or alveolar). The results show that these vowels produced in context exhibit the patterns Macmillan *et al.* (1988) discovered for consonants, consistent with the idea that both vowels in context and consonants are perceived as linguistic entities, whereas isolated vowels are not. The results also show a difference in perception between the two consonant contexts. For vowels produced in the labial context, the boundary between front versus back perception was closer to the /u/ end of the continuum than for vowels produced in the alveolar context. This reflects compensation by listeners for the expected effects of coarticulation with the flanking consonants [Ohala and Feder, Phonetica **51**, 111–118 (1994); Bradlow and Kingston, J. Acoust. Soc. Am. Suppl. 1 **88**, S56 (1990)]. [Work supported by NIH Grant No. 5-R29-DC01708.]

5aSC16. Vowel discrimination in a phonologically neutralized context. Robert Allen Fox and Son Tran (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

The perception of vowel quality is not only affected by the formant structure of the vowel, but by the nature of the surrounding phonetic context. For example, it has been demonstrated [R. A. Fox, in *Speech Perception, Production and Linguistic Structure*, edited by Y. Tohkura *et al.* (IOS Press, 1992)] that listeners' ability to make reliable vowel quality distinctions are lowered when the vowels are placed in a context that introduces phonological neutralization (e.g., post-vocalic [ɹ]). One possible explanation for this phenomenon is that of listener bias, which should be eliminated using the appropriate signal detection methodology. The present study describes the results of an experiment that obtained d' measurements of one-step differences in an [i]-[ɪ] continuum in CV, CVr, CVl, and CVd contexts under fixed-level, minimum uncertainty conditions. Results demonstrate after several days experience with the stimuli and methodology that the discrimination differences as a function of context were eliminated. However, a post-training identification test still showed context effects on vowel categorization. These data will be discussed in terms of the possible nature of this listener bias in vowel perception.

5aSC17. Temporal and acoustic parameters mediating perceptual overshoot in vowel perception. John W. Hawks (Kent State Univ., School of Speech Pathol. and Audiol., Kent, OH 44242)

The experiments reported here represent the preliminary findings of an investigation designed to provide a better understanding of the parameters underlying perceptual overshoot in vowel perception. Continua of dynamic vocalic tokens were synthesized which spanned multiple vowel categories and varied in (1) $F1$, $F2$ location of starting and ending points, (2) rate of frequency change, (3) direction of dynamic transition, and (4) total duration. Subjects determined (1) whether each token was heard as containing one or more than one vowel quality, and (2) the identity of the vowel quality(s). The results suggest the values of the varied parameters sufficient for eliciting the overshoot phenomenon and will be discussed in terms of evidence for a purely auditory mechanism for overshoot or whether an articulatory (biomechanical) basis can be considered.

5aSC18. The perceptual magnet effect as an emergent feature of neural map formation. Frank H. Guenther and Marin N. Gjaja (Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215)

The well-known perceptual magnet effect [e.g., P. K. Kuhl, Percept. Psychophys. **50**, 93–107 (1991)] is characterized by a warping of perceptual space near phonemic category centers. Whereas previous explanations of this effect have been formulated within the theoretical framework of cognitive psychology, the model proposed here builds on research from both psychology and neuroscience to account for the effect. This model embodies two principal hypotheses supported by considerable experimental and theoretical research from the neuroscience literature: (1) Sensory experience guides development of an auditory neural map, and (2) a population vector analysis can predict psychological phenomena from map cell activities. These hypotheses are realized in a simple, self-organizing neural network model. The magnet effect arises in the model from language-specific nonuniformities in the distribution of map cell firing preferences. These nonuniformities result from exposure to language-specific distributions of attended sounds experienced by the network during training, in keeping with neurophysiological results from auditory cortex [G. H. Recanzone, C. E. Schreiner, and M. M. Merzenich, J. Neurosci. **13**, 87–103 (1993)] and other sensory areas. Numerical simulations verify that the model captures the known general characteristics of the magnet effect and provides very accurate fits to specific psychophysical data.

5aSC19. The phonological relationship between Taiwanese initial voiced stops and nasals. Ho-hsien Pan (Dept. of Foreign Languages and Literature, Chiao Tung Univ., Hsinchu, 30050, Taiwan)

Taiwanese voiced stops are followed by non-nasalized vowels, while nasals are followed by nasalized vowels. This study uses a concept formation paradigm to test whether native speakers treat the voiced stops and nasals as allophones or separate phonemes. There are three experiments in the study. In the training session of experiment A, monosyllabic morphemes are presented. Subjects receive a positive reinforcement when a syllable beginning with /b/ is presented. In the test session of experiment A, syllables beginning with /m/ are introduced. In experiments B and C disyllabic phrases are presented. In the training session of test B, the subjects are reinforced to treat /b/ and /m/ as separate categories. In the training session of test C, the subjects are reinforced to treat /b/ and /m/ as the same category. The results show that /m,b/ are grouped into the same category. In experiment B, subjects cannot differentiate between /b/ and /m/. In experiment C, when /b/ and /m/ are reinforced to be of the same category, the subjects perform well. According to the interview responses after each test, subjects describe the category that is being reinforced positively as nasals.

5aSC20. Modeling processing dependences in speech perception. Court S. Crowther (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

The goal of this work was to determine whether acoustic cues to the phonetic features of a segment are processed independently. For stop–vowel (CV) syllables, one point of controversy concerns whether place information (as conveyed by formant transitions) and voicing information (as conveyed by VOT) are processed mutually independently. In this experiment, a replication of Massaro and Oden [J. Acoust. Soc. Am. **67**, 996–1013 (1980)], $F2$ and $F3$ transitions and VOT were manipulated as place and voicing cues in CVs. Members of a family of multinomial models, including FLMP, were fitted to each subject's data. FLMP, which assumes that processing is independent at both the perceptual and decisional levels, failed to account for the data. Two other models retained the perceptual independence assumption, but one assumed the voicing decision was contingent on the outcome of the place decision, and the other assumed the opposite contingency. Both fitted better than FLMP, but nevertheless were rejected on statistical grounds. A model assuming decisional independence, but perceptual dependence, fitted each subject's data well. It was concluded that the formant transitions and VOT were not perceived independently, but that the voicing and place decisions were made independently. [Work supported by NIDCD (2-T32-DC-00013-16).]

5aSC21. Effects of amplitude on voicing contrast may not be explained by VIIIth nerve synchrony capture. Andrew J. Lotto and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706)

Kluender, Lotto, and Jenison [J. Acoust. Soc. Am. **97**, 2552–2567 (1995)] reported that overall stimulus amplitude can affect perception of the voicing contrast in syllable-initial stops. Their results supported an hypothesis that a shift in the temporal pattern of neural firing from frequencies near $F2$ and $F3$ to $F1$ and $f0$ could signal voicelessness. The results of the current study undermine this "synchrony capture hypothesis." The effect of amplitude (increased voiceless identifications with higher amplitude) maintains when there is no cutback in $F1$ during the quasiperiodic portion of the syllable and when stimuli are high-pass filtered above the frequency of $F1$. In a further test of the hypothesis, two ten-step series (/ba/–/pa/ and /ga/–/ka/) were created which maintained period voicing throughout the syllable (with $F1$ cutback signaling voicelessness). The energy just below the frequency of $F2$ and the energy above $F1$ were presented dichotically. Thus, at the periphery, there was no competition between frequencies near $F2$ and lower frequencies and, as a result, no chance for a change in neural temporal patterns. Subjects continued to label voicelessness as a function of overall amplitude. Alternative models of the encoding of voicelessness will be considered. [Work supported by NIDCD and NSF.]

5aSC22. Auditory representation of changes in vocal dynamic level. Shari L. Campbell (Dept. of Commun. Sci. and Disord., Univ. of Georgia, Athens, GA 30602-7152)

It is well known that changes in vocal dynamic level do not result in a simple loudness increase. Rather, because the level of the higher harmonics tends to increase more rapidly than that of lower harmonics, distinct changes in timbre typically accompany the obvious changes in loudness. However, without reference to auditory transformations, it is difficult to determine which of the observed spectral changes are preserved in the "auditory spectrum," or to estimate their perceptual effects. In order to address these issues, sung vowels were recorded from several classical singers. Data represent six voice classifications, three vowels, three dynamic levels, and three points within each singer's fundamental frequency range. Various unidimensional metrics based on models of auditory processes such as frequency selectivity, masking, loudness perception, and timbre perception [Glasberg and Moore, Hear. Res. **47**, 103–138 (1990); Aures, Acustica **59**, 130–141 (1985)] were obtained. Results to date indicate that the weighted first moments of the specific loudness and sharpness distributions serve as promising means of quantifying changes associated with changes in vocal dynamic level, and that these metrics are most effective when derived from excitation patterns based on auditory filter shapes rather than masking patterns. [Work supported by UGACOE Faculty Research Grant.]

5aSC23. A novel psychophysical procedure for investigating the syllabic organization of phonemes in speech. James A. Bashford, Jr., Richard M. Warren, and Bradley S. Brubaker (Psych. Dept., Univ. of Wisconsin, Milwaukee, WI 53201)

When listeners attempted to tap in synchrony with a target phoneme in a repeating phrase or sentence (e.g., "Most Wisconsin winters are cold."), taps to syllable-initial phonemes were accurate, whereas taps to syllable-final phonemes were early. These results indicate that the apparent occurrence of speech sounds depends upon syllabification, and that the synchronous phoneme-tapping task can be used for investigating syllabic structure. The tapping paradigm was used to examine a topic of current interest—the syllabic assignment of intervocalic consonants. The stimuli were six minimal-contrast pairs of bisyllabic CVCVC words differing only in their first vowels, having either a tense V1 (as in "tunic") or a lax V1 (as in "tonic"). Tense-V1 and lax-V1 words were presented to separate groups of 24 listeners who provided a sequence of 15 taps to each target phoneme in the repeated words. Intervocalic consonants (IVCs) following tense vowels elicited accurate tapping, indicating their treatment as second-syllable onsets, whereas taps to IVCs following lax vowels were early, indicating their treatment as first-syllable codas. These results are consistent with linguistic theory and explicit judgments of syllabification [see Trieman and Danis, J. Mem. Lang. **27**, 87–104 (1988)].

5aSC24. First formant spectral properties and initial stop–consonant [voice] judgments. Michelle R. Molis and Randy L. Diehl (Dept. of Psych., Univ. of Texas, Austin, TX 78712)

A low first formant ($F1$) onset frequency is an acoustic correlate of [+voice] consonants. This is predicted by the "low-frequency hypothesis" (LFH)—any increase in low-frequency energy near the consonantal closure will enhance the perception of [+voice] stop consonants. In contrast, Kluender, Lotto, and Jenison [J. Acoust. Soc. Am. **97**, 2552–2567 (1995)] proposed that $F1$ onset frequency acts as a cue via an indirect effect on the temporal firing pattern of auditory-nerve fibers. Following voice onset, the synchronous firing of some mid- and high-CF fibers may become captured by energy in the region of $F1$. The degree of capture grows with decreased $F1/F2$ distance. It can also be enhanced by increasing the relative intensity of $F1$. In cases where $F1$ is more intense, the LFH predicts more [+voice] responses because the absolute amount of low-frequency energy is greater; the "synchrony capture hypothesis" predicts fewer [+voice] responses because synchrony capture is more likely as $F1$ intensity grows. In fact, manipulation of $F1$ intensity produced no change in subjects' responses indicating that neither hypothesis is adequate in its present formulation. Evidence for changes in synchrony capture is provided by two computer models of auditory-nerve function. [Work supported by NIH.]

5aSC25. Preschool children's use of dynamic acoustic information for word recognition. Benjamin Munson, Jan Edwards, and Robert Allen Fox (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

While young children's ability to use dynamic acoustic information to categorize speech sounds has been studied in some detail [Nittrouer, J. Phon. **20**, 1–32 (1992)], relatively little is known about their ability to use dynamic information for word recognition. This study investigates whether preschool children could use dynamic acoustic information to identify familiar words with missing consonants or vowels. A gated series [Elliot *et al.*, Percept. Psychophys. **42**, 150–157 (1987)] was used to examine young children's identification of familiar CVC words with missing final consonants; similarly, a silent center series [Fox *et al.*, J. Speech Hear. Res. **35**, 892–902 (1992)] was used to examine identification of CVC words with missing vowels. Eighteen typically developing children aged three to five were tested. It was found that even the youngest children did remarkably well at recognizing words with missing final consonants, even at the longest gate when the consonant was deleted 40 ms before stop closure. By contrast, the children had more difficulty with the silent center series, with some children being unable to perform the task. Results will be presented also for a second set of gating and silent center tasks, in which a picture-pointing response, rather than a spoken response, was used.

5aSC26. Auditory-visual context effects on the perception of /r/ and /l/ in a stop cluster. Linda W. Norrix (Ctr. for Neurogenic and Commun. Disord., Univ. of Arizona, Tucson, AZ 85721) and Kerry P. Green (Univ. of Arizona, Tucson, AZ 85721)

Context effects in speech perception are thought to reflect knowledge about coarticulatory influences from the surrounding phonetic environment. This study investigated if such effects occur when the context is specified in the visual modality and segmental information in the auditory modality. In the first experiment, two continua were synthesized: one varying from /iri–ili/ and the other from /ibri–ibli/. When presented to listeners for identification as /r/ or /l/, there was a significant shift in the category boundary between the two continua. In a second experiment, the /iri–ili/ tokens were paired with visual tokens of a talker saying /ibi/. When presented in an auditory-visual (AV) condition, these tokens were perceived as ranging from /ibri/ to /ibli/. Observers also identified the /iri–ili/ tokens in a separate auditory-only (AO) condition. Results indicated a similar shift in the /r–l/ boundary between the AO and AV conditions. Analysis of /r/ and /l/ productions revealed that the perceptual adjustments were consistent with the acoustic consequences of articulating /r/ and /l/ in a stop cluster. The findings suggest that the perceptual adjustment reflects cross-modal knowledge of the coarticulatory effects of a bilabial consonant on the acoustic realization of /r/ and /l/. [Work supported by NIDCD, NIH.]

5aSC27. A reconsideration of intensity-based duplex perception for speech. Michael D. Hall and Patricia K. Kuhl (Dept. of Speech and Hear. Sci., CHDD, Box 357920, Univ. of Washington, Seattle, WA 98195-7920)

Duplex perception (DP) occurs when one stimulus component simultaneously contributes to two distinct percepts. In one potential demonstration, a sinusoidal glide is substituted for an $F3$ transition in a /da/ or /ga/ syllable, and the glide is claimed to distinguish the consonant while simultaneously being separately perceived as a nonspeech chirp. Recent concerns regarding the validity of evidence for this variant of DP were addressed using theoretically bias-free psychophysical methods. Individual thresholds for consonant, and chirp, perception were compared as a function of glide intensity. Consistent with DP, consonant identification and discrimination were maintained at glide intensities well below chirp detection threshold. Listeners also could not reliably match isolated glides to glides in speech context, indicating that consonants were not simply inferred from the perception of isolated glides. Methodological implications of the results are discussed, including the need to insure that the nature of consonant perception depends upon the mixture of a given glide with the remainder of the syllable. An alternative, and potentially more reliable method for evaluating chirp detection threshold also is provided. [Work supported by NIH.]

5aSC28. Individual differences and the perception-production link. Rochelle S. Newman (Dept. of Psych., Park Hall, SUNY, Buffalo, NY 14260)

Several theories (e.g., Motor Theory [Liberman and Mattingly, Cognition **21**, 1– 36, 1985]) posit a link between perception and production. One way of examining the interrelationship between production and perception is to look at individual differences in both types of tasks. Previous research has yielded somewhat ambiguous results, but many of these studies looked at the boundaries between categories. The current study examined individuals' prototypes and related data from production. Subjects participated in a perceptual task modeled on that of Miller and Volaitis [Percept. Psychophys. **46**, 505–512 1989]. Subjects heard members of a VOT series centered on /pa/, and rated them according to "p" category goodness. These same subjects were recorded producing stop consonant–vowel tokens, and the VOTs of these productions were measured. A correlation was found between the average VOT for subjects' productions of "pa" and the VOT of the tokens they rated as the best examples. This supports the idea of a perception-production link, and leaves open the possibility that a common mechanism may be used in both processes. [Work supported by NIDCD Grant No. R01 DC00219 to SUNY at Buffalo.]

5aSC29. Stop consonant perception in 3- and 4-year-old children. Ralph N. Ohde and Katarina L. Haley (Div. of Hearing and Speech Sciences, Station 17, Vanderbilt Univ. School of Medicine, Nashville, TN 37232)

The purpose of this study was to examine the importance of different acoustic properties for the perception of place of articulation in young children and adults. Ten adults and ten children in both of the age groups, 3 and 4 years, listened to synthesized consonant–vowel syllables comprised of all combinations of [b d g] and [i]. The synthesis parameters included manipulations of the following stimulus variables: formant transition (moving or straight), noise burst (present or absent), and voicing duration (10 or 46 ms). In a recent study [R. N. Ohde *et al.*, J. Acoust. Soc. Am. **97**, 3800–3812 (1995)] of children between 5 and 11 years of age, there was no difference between children and adults in terms of the perceptual weight placed on the dynamic formant transition cue. However, in the current study, preliminary findings indicate that young children's identification of stop consonants was higher for moving transitions than straight transitions. In addition, the children's identification of stops was frequently poor in the [i] context. Overall, these results suggest that children pay particular attention to dynamic cues in the early period of sound acquisition, and that they have reduced auditory sensitivity for acoustic correlates of place of articulation in the [i] context. [Work supported by NIDCD.]

5aSC30. Age-related changes in the categorical perception of stop consonants. Elzbieta B. Slawinski (Dept. of Psych., Univ. of Calgary, 2500 University Dr., Calgary, AB T2N 1N4, Canada)

The study focused on changes in the categorical perception of intial stop consonants as a function of age and hearing status. The identification of intial stop consonants differing in voicing and place of articulation relies on an integration of multiple acoustic cues (VOT, character of the noise burst, onset frequency and direction of the second and third formant transitions). Four hundred subjects, divided into four age groups (20–85 years) and into three hearing groups, participated in experiments that required the identification of syllables [ba] or [pa] and [ba], [da], or [ga] along synthesized continua. Stimuli were presented to subjects via headphones at 75 dB SPL. Results demonstrated that the phonemic boundary of the [ba–pa] continuum was shifted towards a longer VOT as a function of age and hearing status. Moreover, the phonemic boundaries between [ba] and [da], and between [da] and [ga], demonstrated by older and hearing impaired subjects were shifted toward [d] and [g] categories as compared to young subjects with normal hearing. The observed changes may be due to the age-related deterioration of ability to analyze brief signals and ability to integrate multiple acoustical cues. [Work supported by Health and Welfare of Canada.]

5aSC31. An age-based auditory Stroop effect. James V. Ralston, Jarad M. Plesser (Dept. of Psych., Ithaca College, 1119 Williams Hall, Ithaca, NY 14850), and Elizabeth A. Lawless (Ithaca College, Ithaca, NY 14850)

The present research investigates the effect of semantic content of an utterance on age perception. On each trial of two experiments, listeners heard either a relatively old or young voice uttering a word or nonword. Listeners rapidly decided whether the speaker was relatively young or old. One experiment included five repetitions of the words "young" and "old," and a nonword spoken by a college-aged and a middle-aged female. Analyses of reaction times for correct responses revealed a Stroop effect: Responses were quicker when the semantic information and the age of the speaker were consistent than when they were inconsistent. Overall reaction times, but not the Stroop effect, decreased across repetitions. A second experiment utilized the words "young" and "old," age-associates, age-neutral words, and nonwords, all recorded from a college-aged and a middle-aged female. Analyses of reaction times revealed a Stroop effect for the semantic associates. In addition, there was a strong trend for response times for nonwords to be greater than those for neutral words. The combined results demonstrate that lexical information influences a listener's perception of age. Results from more recent studies manipulating listener set will also be presented.

5aSC32. Dynamic cues in vowel identification: A training study. Amy T. Neel and Diane Kewley-Port (Dept. of Speech and Hearing Sciences, Indiana Univ., Bloomington, IN 47405)

The importance of dynamic formant information for vowel identification has been shown by several studies in recent years. Using sine-wave vowel analogs, Neel and Kewley-Port [J. Acoust. Soc. Am. **96**, 3284(A) (1994)] demonstrated that vowel duration is an important cue for vowel identity and that dynamic formant information is more effective when duration cues are absent. However, because identification rates for sine-wave stimuli were low, a training study was conducted to determine the impact of training on the effectiveness of dynamic formant and duration cues for vowel identification. Sine-wave stimuli consisted of two tones representing F1 and F2 from 10 vowels produced by a female speaker. Four types of stimuli were constructed by varying two factors: (1) dynamic versus static formants and (2) appropriate versus fixed vowel duration. Listeners were trained to criterion on one set of stimuli and were tested on another. Training significantly improved identification accuracy. In comparison to performance on static formant tokens, listeners were more accurate in identification and improved more from baseline to test when dynamic formant cues were present. Listeners also showed an identification advantage when duration cues were available. [Research supported by NIHDCD Research Grant No. DC-02229.]

FRIDAY AFTERNOON, 17 MAY 1996 — REGENCY C, 1:15 TO 4:45 P.M.

Session 5pPA

Physical Acoustics: Resonant Ultrasound Spectroscopy II

Hassell M. Ledbetter, Chair
National Institute of Standards Technology, Materials Reliability Division, 325 Broadway, Boulder, Colorado 80303

Invited Papers

1:15

5pPA1. Unusual elastic properties of pure and doped La_2CuO_4 as probed by resonant ultrasound spectroscopy. John L. Sarrao, Z. Fisk (National High Magnetic Field Lab., 1800 E. Paul Dirac Dr., Tallahassee, FL 32306), T. Darling, and A. Migliori (Los Alamos Natl. Lab., Los Alamos, NM)

La_2CuO_4 undergoes a structural phase transition from tetragonal to orthorhombic crystal symmetry as it is cooled below 525 K. Hole-doping the material (for example, by substituting Sr for La or Li for Cu) rapidly suppresses the structural transition temperature. Additionally, the in-plane lattice constant of $La_{2-x}Sr_xCu_{1-y}Li_yO_4$ is a function only of the combined hole count (i.e., $x+y$) and not of the individual Sr and Li concentrations. RUS measurements for stoichiometric, Sr-doped, and Li-doped La_2CuO_4 have been used not only to map out the phase diagram of this system but also to reveal the unconventional nature of the transition. These results suggest a strong coupling between hole concentration and structural properties as well as a large in-plane/out-of-plane anisotropy.

1:45

5pPA2. Cracks in steel rollers: Detection using resonant ultrasound spectroscopy. Ming Lei (Quatrosonics, Inc., 4209 Balloon Rd. NE, Albuquerque, NM 87109)

A resonant ultrasound technique has been applied to detect cracks in steel spherical rollers. Combining theoretical calculations with RUS measurements, steel rollers provided by a commercial source with various dimensions and aspect ratios were studied. As a result, correlations between aspect ratio and deformation type for some resonant modes were established. Empirical relationships between some specific modes and dimensions were also obtained, with which resonant frequencies of torsional, disk bending, and cylinder bending modes can be predicted from dimensions of the rollers for various aspect ratios. Hence, the testing frequency ranges for a given roller can be determined depending on the type of defects that need to be detected.

2:15

5pPA3. An investigation of computational problems associated with resonant ultrasound spectroscopy. P. S. Spoor, P. J. White, and J. D. Maynard (Dept. of Phys., Penn State Univ., University Park, PA 16802)

At a previous meeting (June 1995) a report was given on anomalous behavior of the Rayleigh–Ritz method for calculating the normal mode frequencies of elastic solids (the method that forms the basis for most resonant ultrasound work), when those solids had the shapes of slightly perturbed parallelepipeds. These shapes are of interest to understand the effects of sample preparation errors on the overall elastic constant determination. In an effort to understand these effects, a systematic study of the various interrelated computational issues in RUS has been undertaken. Attempts are made to address the following issues: (1) whether the anomalous behavior of the Rayleigh-Ritz solutions is algorithmic or numerical; (2) whether these anomalies may result from problems with the inverse calculation (where the frequencies are inverted to obtain elastic constants) as well; and (3) what relationship exists between the RMS error in the frequencies and the confidence in the elastic constant determination. Results will be shown comparing analyses of actual data with and without the inclusion of a correction for nonparallelism/nonperpendicularity in the sample. [Work was supported in part by the Office of Naval Research.]

2:45–3:00 Break

3:00

5pPA4. Scattering pole resonances: Relation to external and internal surface waves. H. Überall (Phys. Dept., Catholic Univ., Washington, DC 20064, LAUE, Univ. of Le Havre, 76610 Le Havre, France, and CNRS, Lab. Mécanique et Acoust., 13402 Marseilles, France), J. Duclos, M. El Hocine Khelil, G. Maze, J. Ripoche (Univ. of Le Havre, 76610 Le Havre, France), A. Gérard (Univ. of Bordeaux, 33405 Talence, France), X. L. Bao, P. K. Raju (Auburn Univ., Auburn, AL 36849), J. P. Sessarego, and J. Sageloli (CNRS, 13402 Marseilles, France)

The resonance scattering theory and the singularity expansion method are based on the complex-frequency poles of the scattering amplitude; these are located off the real-frequency axis and lead to prominent resonances. Their physical origin lies in a phase matching of surface waves generated on the scattering object; hence a study of the resonances will lead to information on these surface waves. This will be demonstrated here for immersed elastic objects of plane, spherical, and cylindrical geometry where even the physical nature (fluidborne or elastic type) of the surface waves can be identified along portions of their dispersion curves, and where for multilayered structures individual-layer resonances can be distinguished, leading to solutions of the corresponding inverse problem.

3:15

5pPA5. Radiation impedance of RUS modes in fused silica and KCl. J. Herro, H. Zhang, C. Hucho, D. Beck, M. Levy (Dept. of Phys., Univ. of Wisconsin, 1900 E. Kenwood Blvd., Milwaukee, WI 53211), D. Isaak, J. D. Carnes, and O. Anderson (UCLA, Los Angeles, CA)

The pressure dependence of the resonance frequency of several RUS modes in samples of fused silica and KCl has been measured at UCLA in atmospheres of air, He, and Ar near ambient temperature. For both compressional and torsional modes the radiation resistance appears to be linearly dependent upon pressure and increases with the molecular mass of the surrounding gas. The effects are larger for breathing modes than for torsional modes. A model will be presented which will attempt to fit some of these data both qualitatively and quantitatively. [Work supported by the Office of Naval Research.]

3:30

5pPA6. Electromagnetic excitation and acoustic spectroscopy of the quadrupole elastic mode of an empty nearly spherical shell. Brian T. Hefner and Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

The lowest quadrupole mode of an empty nearly spherical metallic shell was excited by a novel form of EMAT (electromagnetic acoustic transducer) not having mechanical contact with the shell. An oscillating current is supplied to a coil located near the equator of the shell. The shell vibrates in response to the Maxwell stresses which are distributed over the shell's surface in a way especially favorable for excitation of the lowest quadrupole mode (a bending mode) which has a natural frequency near 64 kHz. The stresses are associated with the Lorentz forces on eddy currents around the equator. The mode is detected with a small microphone placed near the shell. In one version of the experiment a large dc bias magnetic field is superposed on a weaker oscillating field of frequency near 64 kHz. In another version of the experiment the bias field is removed and the frequency of the oscillating field is near 32 kHz (half of the excited mechanical frequency). The method may have applications to the inference of the contents or elastic properties of shells and to the investigation of the modal structure of open shells. [Work supported by the Office of Naval Research.]

3:45

5pPA7. Resonant ultrasound in circular pipes. Fred M. Mueller, Dipen N. Sinha, Roger D. Hasse, and Kendall N. Springer (Los Alamos Natl. Lab., Los Alamos, NM 87545)

Resonant ultrasound has found multiple applications, ranging from medical imaging to a variety of nondestructive testing techniques. Here focus is placed on nested cylinders. A series of ultrasonic measurements has been carried out in a geometry with an outer hollow steel cylinder, and a smaller, solid inner steel cylinder. In the interspace were placed several liquids. Ultrasound was introduced by a narrow contact transducer. A second set of narrow transducers was used as pickups and placed at angles ranging from 10 to 150 deg. The frequency of the ultrasound was varied from 1 to 5 MHz and showed a sequence of eight composite and sharp resonances spaced at about 0.5 MHz. The composite resonances showed a strongly skewed behavior as a function of frequency. The velocity potential has been modeled by using combinations of Hankel functions of complex argument. Convergence was achieved by using angular m's up to 150. These showed that the asymmetry of the composite resonance peaks is primarily a phase or geometrical optics effect.

4:00

5pPA8. Free vibrations of $LiNbO_3$ piezoelectric crystals. Martin L. Dunn (Dept. of Mech. Eng., Univ. of Colorado, Boulder, CO 80309-0427), Hassel Ledbetter (National Inst. of Standards and Technol., Boulder, CO 80303), and Paul Heyliger (Colorado State Univ., Fort Collins, CO 80523)

The first steps toward measuring the elastic and piezoelectric constants of piezoelectric solids using resonant ultrasound spectroscopy (RUS) have been undertaken. Specifically, free-vibration frequencies of a piezoelectric $LiNbO_3$ crystal were measured up to 1 MHz. The vibration frequencies of the piezoelectric parallelepiped were also predicted using the finite element method with published values of the elastic, piezoelectric, and dielectric constants of $LiNbO_3$. Published values of these constants from different sources using different methods are in reasonably good agreement. They also agree well with measurements of some of the constants performed in this study using a pulse-echo technique. Predicted piezoelectric vibration frequencies agree well with measured frequencies. Predictions from an elastic vibration analysis underestimate those from a piezoelectric vibration analysis by up to 12%. The results presented here suggest the feasibility of developing an inversion method to extract both the elastic and piezoelectric constants. This would extend the study of Ohno [I. Ohno, Phys. Chem. Minerals **17**, 371–378 (1990)] who used RUS to determine the elastic constants of alpha quartz. The piezoelectric effect was considered, but no attempt was made to determine the piezoelectric constants as they were held fixed during the inversion to obtain the elastic constants.

4:15

5pPA9. The inherent background in acoustic wave scattering by a submerged target. Myoung Seon Choi, Young Sang Joo, and Jong Po Lee (Nondestruct. Eval. Team, Korea Atomic Energy Res. Inst., P.O. Box 105, Yusong, Taejon 305-600, Korea)

A new concept for the background in resonance scattering of acoustic waves by a submerged target is proposed. The idea is that the remainder after removing only the resonances from the target response must be used as the background. The complex wave numbers whose imaginary parts are the attenuation coefficients of elastic bulk waves in target are introduced to damp out the resonances. From the assumption that the structural damping coefficient, the product of the attenuation coefficients of elastic bulk waves, and the outer radius of target, is a large constant compared with the radiation damping coefficients of resonances, a simple mathematical expression is derived for elastic cylindrical shells and named the inherent

background. It is shown that the inherent background leads to the impenetrable (i.e., rigid or soft) background in some limiting cases and also describes exactly the background of elastic shell over entire frequencies regardless of shell thickness.

4:30

5pPA10. RUS investigation of the anisotropic elastic properties of GFRP composites at low temperatures. Timothy M. Whitney and Robert E. Green, Jr. (Ctr. for Nondestruct. Eval., Johns Hopkins Univ., Baltimore, MD 21218)

The resonance spectrum of any specimen is dependent on its symmetry, density, geometry, elastic properties, and the boundary conditions. By using specimens with simple known geometry and models including specimen symmetry, resonance ultrasound spectrosopy can determine the full elastic stiffness tensor to great precision with one measurement. This measurement is fast, taking less than a minute with state-of-the-art instrumentation, making it appropriate for measuring properties as a function of temperature. Parts per million changes in specimen density, geometry, elastic moduli, temperature, and boundary conditions can be detected in carbon fiber reinforced epoxy composite materials with resonance ultrasound spectroscopy. It is important to know the mechanical properties of carbon fiber reinforced epoxy composite materials at low temperatures for underwater, space, and polar applications. The amplitude/frequency resonance spectra of four different lay-ups of AS4/3501-6 carbon fiber reinforced epoxy composite have been measured at 1 °C intervals from −177 °C to 25 °C. The spectra indicate increasing stiffness and decreasing damping with decreasing temperature.

FRIDAY AFTERNOON, 17 MAY 1996 MT. RAINIER AND MT. MCKINLEY, 1:00 TO 4:00 P.M.

Session 5pPP

Psychological and Physiological Acoustics: Auditory Localization and Pattern Perception (Poster Session)

Robert A. Lutfi, Chair

Department of Communicative Disorders, Waisman Center, University of Wisconsin, 1500 Highland Avenue, Madison, Wisconsin 53705

Contributed Papers

All posters will be on display from 1:00 to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 to 4:00 p.m.

5pPP1. The masking-level difference in low-noise noise. Joseph W. Hall III, John H. Grose (Div. Otolaryngol., Univ. North Carolina at Chapel Hill, Chapel Hill, NC 27599-7070), and William M. Hartmann (Michigan State Univ., East Lansing, MI 48824)

The masking-level difference for a 500-Hz pure tone signal was examined for an interaurally in-phase noise having different degrees of power fluctuations. MLDs were reliably smaller in noise having relatively low power fluctuations. This effect was due to the following trends in the threshold data. (1) For all subjects, the So threshold improved as the degree of noise fluctuation decreased. (2) For some subjects, the $S\pi$ threshold worsened as the degree of noise fluctuation decreased. (3) For some subjects, the $S\pi$ threshold stayed constant as the degree of noise fluctuation decreased. These results are discussed in terms of comodulation masking release and the relevance of envelope cues for the MLD. [Work supported by AFOSR.]

5pPP2. The masking level difference in children: Effects of signal and masker bandwidths. John H. Grose, Joseph W. Hall III, and Madhu Dev (Div. Otolaryngol./Head & Neck Surgery, Univ. North Carolina, Chapel Hill, NC 27599-7070)

The age at which the magnitude of the masking level difference (MLD) reaches adult levels depends partly upon masker bandwidth (BW). For wider BWs, children perform like adults by 5–6 years of age. For narrow BWs, adult-like performance is not reached until an older age. To examine this age dependence further, this study measured the MLD for a 500-Hz tone as a function of masker BW in a group of 20 children aged 5–10 years, and in a group of 10 adults. Masker BWs were 20, 40, 80, 160, 320, and 1000 Hz, centered at 500 Hz. The MLDs of the children were generally smaller than those of the adults for BW = 20 Hz. Children over 7 years of age had adult-like MLDs for all BWs ≥ 40 Hz. The youngest children did not reach adult levels until BWs ≥ 320 Hz. To determine whether the smaller MLDs at narrow BWs were due to the perceptual similarity between the masker and the signal, MLDs were also measured in conditions where both masker and signal had BWs = 20 or 320 Hz. Children under 8 years of age showed the same MLDs for both BWs. Older children and adults had larger MLDs for the narrow BW. [This work supported by NIH NIDCD.]

5pPP3. Benefit of binaural hearing in a multi-source environment. I. Sound localization. Ruth Y. Litovsky, Monica L. Hawley, Jennifer K. Jones, Leah B. Dunton, and H. Steven Colburn (Dept. of Biomedical Eng. and Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215)

A multi-source listening environment was simulated under headphones by recording signals through KEMAR in a sound-deadened room. Seven loudspeakers were each positioned in the frontal plane 5 ft from KEMAR at 30-deg increments, ranging from +90 (right) to −90 (left). The recorded signals (IEEE Harvard sentences) were digitally mixed and played back under headphones in a sound-proof booth using a Digital Audio Tape (DAT) player. Sound locazation accuracy was measured for "target" sentence in the presence of various "jammers." Three variables were manipulated. First, the number of "jammers" was either none, one, two or three. Second, the relative location of the targets and jammers was varied, such that they were either in close proximity, widely separated, or some of each. Third, the jammer content comprised of either other sentences, speech-shaped noise, babble or time-reversed sentences. Normal-hearing listeners were tested under binaural and monaural conditions. The content of the target sentence was known. Localization accuracy decreased as the number

5p FRI. PM

of jammers increased, and as the proximity of the target and jammers was increased. In addition, performance was worse under monaural than under binaural conditions. [Work supported by NIH DC02696, DC00100-21.]

5pPP4. Benefit of binaural hearing in a multi-source environment. II. Speech intelligibility. Monica L. Hawley, Ruth Y. Litovsky, Leah B. Dunton, Jennifer K. Jones, and H. Steven Colburn (Dept. of Biomedical Eng. and Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215)

A multi-source listening environment was simulated under headphones by recording signals through KEMAR in a sound-deadened room. Seven loudspeakers were each positioned in the frontal plane 5 ft from KEMAR at 30-deg increments, ranging from +90 (right) to −90 (left). The recorded signals (IEEE Harvard sentences) were digitally mixed and played back under headphones in a sound-proof booth using a Digital Audio Tape (DAT) player. Speech intelligibility was measured for a "target" sentence in the presence of various "jammers." Three variables were manipulated. First, the number of "jammers" was either none, one, two, or three. Second, the relative location of the targets and jammers was varied, such that they were either in close proximity, widely separated, or some of each. Third, the jammer content comprised of either other sentences, speech-shaped noise, babble or time-reversed sentences. Normal-hearing listeners were tested under binaural and monaural conditions. On each trial the location of the target sentence was known, and the percentage of key words correctly identified is reported. Speech intelligibility was degraded as the number of jammers and as the target proximity to the jammers was increased. [Work supported by NIH DC02696, DC00100-21.]

5pPP5. The fidelity of virtual auditory displays. Pavel Zahorik, Frederic L. Wightman (Dept. of Psych. and Waisman Ctr., Univ. of Wisconsin, Madison, WI 53705), and Doris J. Kistler (Univ. of Wisconsin, Madison, WI 53705)

A new method for on-line virtual sound source synthesis has been developed that affords direct comparison of virtual and real sound sources. Using this method, four listeners performed a 2AFC discrimination of real free-field sources and the virtual representations of these sources. The sources consisted of wideband noise bursts presented at a number of spatial positions. Virtual source fidelity was manipulated (on a trial by trial basis) by applying windows of various lengths (20.48, 5.12, 1.28, 0.32 ms) to the impulse responses (20.48 ms in duration) of the filters used to synthesize the virtual sources. When fidelity was highest, listeners could not discriminate virtual from real sound sources. As window length was decreased, discriminability increased. For a 1.28-ms window, performance was above 75% correct on average. These results are consistent with physical measurements of the differences in the acoustic signal at the listener's ear drum between virtual and real source presentation. In a related experiment, the same four listeners where asked to report the apparent positions of the virtual sound sources synthesized in the previous experiment. Degradation in localization performance was observed as filter window length was decreased. [Work supported by NASA.]

5pPP6. Correlational analysis of acoustic cues for the discrimination of auditory motion. Wen Wang and Robert A. Lutfi (Psych. Dept., Univ. of Wisconsin, Madison, WI 53706)

Previously, it was reported that differential auditory thresholds for velocity and acceleration do not appear simply related to thresholds for displacement using synthesized sound sources [Wang and Lutfi, J. Acoust. Soc. Am. **95**, 2897(A) (1994)]. The present study was undertaken to determine if the failure to find a simple relation might be due to differences in the types of cues mediating these thresholds. The sound of a source moving in a straight path on the azimuthal plane was synthesized over headphones to include three dynamic cues for motion: Doppler shift, intensity, and interaural delay. The three cues were perturbed slightly from trial to trial so that correlations with listener responses could be used to estimate the relative weight given to each cue. A 2IFC procedure was used to measure differential thresholds for displacement, velocity, and acceleration of four listeners. For the discrimination of displacement, responses were most highly correlated with intensity or interaural delay, but for the discrimination of velocity and acceleration, responses were most highly correlated with Doppler shift. The results provide a means of accounting for differences in sensitivity to motion that cannot be inferred from sensitivity to displacement alone. [Research supported by NIDCD R01 DC01262-04.]

5pPP7. Auditory localization aftereffects. Makio Kashino and Shin'ya Nishida (NTT Basic Res. Labs., 3-1, Morinosato Wakamiya, Atsugi, Kanagawa, 243-01 Japan)

It was observed that the perceptual lateralization of a sound (test) having an interaural time difference (ITD) is shifted away from that of a prior sound (adapter) having a different ITD. First, the frequency selectivity of this "auditory lateralization aftereffect" was examined for sinusoids presented through headphones, with various combinations of adapter and test frequencies below 800 Hz, using the method of constant stimuli. The magnitude of the aftereffect was found to be largest when the frequencies of the two sounds were similar. It decreased as the frequency difference increased, and virtually disappeared at a frequency difference of one-half octave. Second, the ITD selectivity of the aftereffect was examined for 400-Hz tones. Subjects' judgments of lateralization were measured directly in terms of the azimuth of the test signal for various combinations of adapter and test ITDs in the range of ± 625 μs. It was found that the magnitude of the aftereffect was rather small when the ITDs of the adapter and test signal were similar, and was largest when ITDs differed by 250 or 375 μs. These results will be discussed in terms of implications for theories of binaural interaction, possible neural sites, and functional roles of the effects.

5pPP8. One, two, many—Judging the number of concurrent talkers. Makio Kashino and Tatsuya Hirahara (NTT Basic Res. Labs., 3-1, Morinosato Wakamiya, Atsugi, Kanagawa, 243-01 Japan)

The ability of listeners to judge the number of concurrent talkers was examined. Ten female and 11 male Japanese talkers each recorded 20 familiar Japanese words consisting of four consonant–vowel syllables each. In each trial, a number of different talkers was chosen randomly from the same-sex group, and presented synchronously to four native Japanese listeners, who were asked to judge how many talkers were present. The range of talker numbers was unknown to the listeners. To eliminate cues associated with level, the over-all sound pressure level was varied randomly in each trial, with RMS levels of the individual words equalized. It was found that judgments were nearly perfect for up to two talkers, but deteriorated abruptly for three or more talkers. In the latter case, the number of talkers was underestimated, although estimates increased slightly as the number of talkers increased. Factors that may promote sound source separation, such as lexicality (e.g., forward versus reverse speech) and spatial separation (e.g., single versus multiple loudspeakers), did not affect performance. Understanding a message in the presence of other voices may be different from the judgment of number of talkers; the former requires only the separation into two entities—one and the others.

5pPP9. Effect of the number of distracting streams on the focusing of attention on one auditory stream. Renaud Brochard, Carolyn Drake, Steve McAdams, and Marie-Claire Botte (Lab. of Exp. Psych., CNRS URA 316, 28 rue Serpente, 75006 Paris, France)

The experiment reported here deals with the conditions under which listeners are able to focus their attention on one particular auditory stream within multi-stream sequences. Such sequences were composed of two to four isochronous pulses each with a different tempo (ISI=300, 400, 500 or 700 ms) and frequency (from 234 Hz). The effective focusing of attention on one stream was obtained by asking subjects to detect a local temporal irregularity in a target stream in the presence of one, two, or three other streams. To help subjects focus their attention, the multi-streams sequences were preceded by a cue composed of the regular target sequence. Temporal irregularity detection thresholds in the target stream were measured using a 2AFC paradigm (4 up/1 down). The results showed, contrary to expectations, that the ability to focus on one stream was independent of the number of streams (no difference in thresholds when more than two streams were added). So, listeners seem to create two perceptual spaces: one composed of the focused stream, and one composed of the nonfocused streams, irrespective of their physical characteristics. It is also easier to focus on outer streams (the highest or the lowest frequency) than on inner streams.

5pPP10. The influence of grouping and dimensional similarity on interference effects in auditory short-term memory. Gary E. Starr and Mark A. Pitt (Dept. of Psych., Ohio State Univ., 142 Townshend Hall, 1885 Neil Ave., Columbus, OH 43210-1222)

Previous research investigating auditory short-term memory using a tone discrimination paradigm found same-dimension similarity to be the primary source of memory disruption [e.g., C. Semal and L. Demany, J. Acoust. Soc. Am. **89**, 2404–2410 (1991)]. However, grouping principles (e.g., Gestalt principles) also provide a possible explanation for the observed effects, because increases in same dimension similarity may also increase the likelihood of the standard tone grouping with additional tones. This project investigated the contribution of grouping to interference effects in short term memory for the dimensions of pitch and timbre. A modified "captor effect" task was used, with participants discriminating either the pitch or timbre of a standard tone (the sixth tone in an 11-tone sequence) from a comparison tone presented 5 s after the standard. Two variables were orthogonally varied, the same-dimension distance between the standard tone and the captor tones (either close or far), and the grouping structure of the sequence, with structures that either promoted or prevented grouping of the standard tone with the captor tones. Results suggest that grouping contributes to the amount of interference because increases in grouping led to increases in interference.

5pPP11. The effect of a free-field auditory target's horizontal motion on its detectability. Xin Xiao and D. Wesley Grantham (Div. of Hearing and Speech Sciences, Vanderbilt Univ. School of Medicine, and Bill Wilkerson Ctr., 1114 19th Ave. S., Nashville, TN 37212)

Previous work has shown that the motion of an auditory signal, when simulated over earphones by presenting a binaural stimulus with a changing interaural temporal difference, does *not* enhance its detectability relative to that of a stationary signal [Grantham and Luethke, J. Acoust. Soc. Am. **83**, 1117–1123 (1988)]. The present study will compare the detectability of stationary and moving auditory targets presented in free-field, where all cues for location, including interaural level differences and spectral information, will be available. Six normal-hearing subjects will be tested individually in a darkened anechoic chamber. Masked threshold will be determined in an adaptive, single-interval 2AFC procedure for targets that are either stationary or moving at various velocities. Three different signals will be employed: a 500-Hz tone, an 8000-Hz tone, and a broadband noise. Preliminary results from one subject indicate that motion may enhance the detectability of broadband noise targets, but not of the pure-tone targets. [Work supported by NIDCD.]

5pPP12. Effects of musician's earplugs and protective headgear on localization ability in the horizontal plane. Nancy L. Vause and D. Wesley Grantham (Div. of Hearing and Speech Sciences, Vanderbilt Univ. School of Medicine, and Bill Wilkerson Ctr., 1114 19th Ave. S., Nashville, TN 37212)

The purpose of this study is to determine how well normal-hearing humans can localize sound sources while wearing protective headgear with and without hearing protection. Six subjects will participate in a source identification task to be conducted in an anechoic chamber. On each trial the stimulus (a 100-ms broadband source) will be presented from one of 20 loudspeakers arrayed in a semicircular arc, and the subject must state which loudspeaker emitted the sound. The arc spans 160° in the horizontal plane at ear level and is about 1.8 m distant from the subject. Each subject will be tested in eight conditions, involving various combinations of wearing protective headgear and three types of earplugs: the EAR plug and the Etymotic ER15 and ER25 "musician's" earplugs. In addition, testing will be conducted at each of two orientations: frontal (center of the array at subject's midline) and lateral (center of the array at 90° azimuth). Results are expected (a) to reveal whether the Etymotic earplugs (which are designed to attenuate equally across a broad frequency range) preserve localization accuracy, and (b) to quantify the interactive effects on localization performance of protective headgear and earplugs. [Work supported by NIDCD and NOHR.]

5pPP13. Lateralization of nonspeech audio-visual stimulus combinations. N. J. Holt (Dept. of Psych., Univ. of York, Heslington, York YO1 5DD, England)

Ability to lateralize stimuli was measured in eight normally hearing subjects. In experiment 1 auditory or visual stimuli were presented. Subjects responded with an auditory or visual pointer in conditions where stimulus and response modalities were the same (uni-modal) or different (cross-modal). A linear relationship was found between the position of the target stimuli and the perceived lateral position, establishing the correspondence between auditorily and visually presented positions, consistent with Yost [J. Acoust. Soc. Am. **70**, 397–409 (1981)]. Mean judgments of linear position were independent of stimulus or response modality. In experiment 2 subjects were presented with bi-modal audio-visual stimuli with spatially and temporally correspondent modal components and subjects responded with an auditory pointer. Mean judgments of position were similar to those in experiment 1 but standard deviations were significantly smaller for the bi-modal stimuli relative to uni-modal stimuli. Experiments 3 and 4 involved manipulations of the spatial or temporal relationship between modal components of bi-modal stimuli. Whereas the relative importance of the visual modality was confirmed [Colavita, Percept. Psychophys. **15(2)**, 409–412 (1974)] the results of both experiments indicated that perception of the location of an audio-visual stimulus is influenced by information conveyed in both modalities. [Work supported by UKBBSRC.]

5pPP14. Anticipatory perception of object pass-by using changing intensity alone: A psychophysiological study. Terri A. Erwin and Louis G. Tassinary (Environ. Psychophysiology Lab., College of Architecture, Texas A&M Univ., College Station, TX 77845)

Prospective information, that is, present conditions of stimulation that are informative about a future state, e.g., the timing and/or location of an impending contact, is required for coordinating acts of interception and avoidance. In the present study, acoustic simulations of passing objects (at different speeds and distances) were produced by modulating the amplitude of a 1-kHz tone in accordance with the inverse square law of acoustic intensity. The simulations were used to explore listeners' capacity for auditorally anticipating arrival time simply on the basis of intensity change—formulated as a *tau* variable [see Shaw, McGowan, and Turvey, Ecol. Psychol. **3**, 253–261 (1991)]. Subjects indicated either the arrival time of a single passing object or the first arrival within a pair by squeezing a dynamometer. Response accuracy/variability and anticipatory EMG, SCR, and EEG activity were used as indexes to listeners' ongoing awareness of

object position during the simulated approach. Preliminary results indicate highly systematic accuracy and variability, along with the inadequacy of *tau* as a means of formalizing the relevant prospective information supporting anticipation.

5pPP15. Detection of timing deviations in simple and complex rhythms. Ralph Barnes, Mari Reiss Jones, and James Klein (Dept. of Psych., Ohio State Univ., 1885 Neil Ave., Columbus, OH 43210-1222)

This experiment was designed to determine whether or not polyrhythmic structure and attentional set influence auditory stream segregation. In sequences where streaming does not occur listeners are often quite accurate in detecting small time changes, but this is not true when sequences stream. Here a two-alternative forced-choice task was used to examine time discrimination within polyrhythmic patterns involving two different tone frequencies (high, low). Polyrhythms with narrow, medium, and wide frequency separations between high and low tones were presented to each subject; single frequency control patterns were also used. Listeners determined whether a small time change in the rhythm occurred early or late in a recurrent presentation of each pattern. In addition to manipulation of frequency separation, rhythmic structure (simple, complex) and attentional set (selective, divided) were varied. Results indicated that both frequency separation (narrow, medium, wide) and rhythm (simple, complex) affected time-change detection. These findings have implications for theories of attention and stream segregation. [Work supported by NSF.]

5pPP16. Dichotic temporal order thresholds. Richard E. Pastore and Edward J. Crawley (Dept. of Psych., Binghamton Univ., Binghamton, NY 13905)

Temporal order thresholds (TOT) were measured for dichotically presented stimuli. Earlier research, using diotic or monaural presentation, required subjects to respond to the perceived spectral properties of the earlier onset stimulus. Dichotic presentation provides an opportunity for the subject to respond to ear with earlier onset, pitch with earlier onset, or both. These alternatives allowed two issues to be evaluated: (1) peripheral (changes in neural patterns) versus central coding and (2) the role of pitch processing in response specification. Contrary to arguments that TOT is the result of low level interactions or changes in peripheral coding, all values for responding based on pitch or ear were within the range of previous findings. Thresholds were largest for responding based solely on pitch presumably because temporally greater information is required to specify pitch than ear. In addition, thresholds were smallest when subjects could use a combination of ear and pitch, possibly reflecting a simpler or less uncertain task.

5pPP17. Perceiving source characteristics from complex sounds. Shannon M. Farrington and Richard E. Pastore (Dept. of Psych., Binghamton Univ., Binghamton, NY 13905)

Other than in speech, there have been very few studies investigating the ability of humans to perceive characteristics of events based upon the sounds emitted. However, there is considerable anecdotal evidence that humans often use complex sounds to accurately perceive events in their environment. To investigate this ability, it is important to determine the extent of subjects' ability to map sounds onto certain event characteristics and then to determine the specific acoustic properties which tend to be used in this mapping process. Once this process has been determined, techniques can be developed to improve this ability. Previous research from our laboratory [X. Li, R. J. Logan, and R. E. Pastore, J. Acoust. Soc. Am. **90**, 3036–3049 (1991)] indicated that subjects are able to identify walker gender based upon acoustic properties of four footsteps on a single surface. The current research investigated subjects' ability to identify specific walkers across different surfaces. Results indicated that, to varying degrees, subjects could reliably identify walkers across different surfaces and could generalize this training to a novel task. Further experiments provide a more detailed analysis of the properties of the acoustic signal which reliably map on to subjects ability to identify a specific walker.

5pPP18. Effects of variable pattern context on temporal acuity. Dana Apfelblat and Mari Riess Jones (Dept. of Psych., Ohio State Univ., 1885 Neil Ave., Columbus, OH 43210-1222)

This experiment varied the time pattern of 25 tones sequences to determine its effect on subjects' abilities to discriminate small time changes within one of two isochronous test regions. A two-alternative forced-choice procedure was used in which a small time change ($\pm\Delta t = 21, 39$, or 57 ms) occurred equally often in one of the two test regions. Test regions were distinguished from surrounding context regions, where timing was variable, by higher frequency tones. Both the absolute amount of temporal variability (small, large) and the nature of temporal variability (rate-accelerating, rate-decelerating, and rate-alternating) within context regions were varied. Results showed that listeners' ability to detect time changes within test regions was poorer with large amounts of contextual variability than with smaller amounts. However, minimal effects of the patterning of tempo (rate) within context regions were found.

5pPP19. Roughness scaling of complex tones. B. John Feng (Scientific Res. Lab., Ford Motor Co., 20000 Rotunda Ave., Dearborn, MI 48121) and Gregory H. Wakefield (Univ. of Michigan, Ann Arbor, MI 48109)

The term roughness has been proposed to describe a sensation elicited by sinusoidal amplitude modulation (SAM) of pure tones and modulated wideband noise [E. Terhardt, Acustica **30**, 201–213 (1974)]. Roughness of SAM tones is known to exhibit a close relationship to modulation depth. It is hypothesized that roughness of complex tones, as scaled against modulation depth of a SAM tone, can be predicted from the magnitude of the envelope spectrum. Subjects were asked to scale complex tones on the dimension of roughness by matching a test stimulus to a reference SAM tone of subjectively equivalent roughness. The test stimuli were drawn from complex tones consisting of a variable number of equal amplitude, harmonically related components. The reference set was 12 SAM tones with modulation depths spanning the range from detection threshold to 100 [Viemeister, J. Acoust. Soc. Am. **88**, 1367–1373 (1990)]. The modulation frequency of the reference set varied across tests conditions according to the minimum component spacing of the tests stimulus. Results of the scaling experiments suggest roughness can be predicted as a function of the narrow-band peaks in the magnitude spectrum of the envelope. [Work supported by Ford Motor Company.]

Session 5pSA

Structural Acoustics and Vibration: Acoustics of Coupled Structures II

Courtney B. Burroughs, Chair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16801

Chair's Introduction—1:00

Invited Papers

1:05

5pSA1. Thermodynamic energy flow, modal models, and statistical energy analysis. Dennis S. Bernstein (Aerospace Eng. Dept., Univ. of Michigan, 1320 Beal St., Ann Arbor, MI 48109-2118)

The acoustics and vibration of coupled structures can be efficiently modeled by means of energy flow theory, usually known as statistical energy analysis (SEA). SEA is based upon relations derived from the interaction of pairs of coupled modes and is applied empirically to coupled structures possessing multiple interacting modes. This talk investigates the accuracy of SEA models by deriving exact energy flow relations based upon modal models. Using white noise disturbances but otherwise deterministic structural models, we present energy flow relations for multiple coupled structures. These relations are formulated in terms of thermodynamic energy, which is a measure of subsystem energy dissipation relative to subsystem disturbance intensity. The resulting energy flow models are compared to SEA models based upon coupled, uncoupled, and blocked subsystem energy. The results are applied to systems of coupled oscillators as well as coupled beams to demonstrate predicted energy flow under both strong and weak coupling.

1:35

5pSA2. The design of a satellite boom with enhanced vibration performance using genetic algorithm techniques. A. J. Keane (Dept. of Mech. Eng., Univ. of Southampton, Highfield, Southampton SO17 1BJ, UK)

This paper examines the application of passive vibration control methods to a simplified satellite boom. It is suggested that such methods may allow an efficient means by which to control structural vibrations. In this research, an initial structure is designed having a regular, repeating geometric pattern. This design is then modified using a genetic algorithm (GA), which is one of a number of recently developed evolutionary computing methods. Here the GA changes the geometry of the design by altering the three-dimensional coordinates of its joints. The aim is to minimize the band-averaged noise transmission along the boom. This paper shows that by altering the geometry in this way, significant improvements in the structure's noise performance can be achieved. While this research is far from complete, the work outlined demonstrates the potential usefulness of this approach for controlling structural vibrations.

2:05

5pSA3. Active control of sound transmission through a structure into a coupled enclosure of arbitrary shape. Ben S. Cazzolato and Colin H. Hansen (Dept. of Mech. Eng., Adelaide Univ., S.A., 5005, Australia)

A numerical procedure for assessing the performance of an active noise and vibration control system for a harmonically excited weakly coupled multimodal vibroacoustic system is presented. The structure and the contiguous acoustic space have been modeled separately with a commercially available finite element analysis package (ANSYS) and then combined using a modal coupling theory for weakly coupled systems. Numerical results are compared against experimental measurements for a finite longitudinally stiffened cylinder with an integral floor. Experimental based response matrices were measured and used to predict the system response. Modal analysis was applied in order to reduce the required effort in determining the response matrix. A design methodology for optimizing the physical control system is reviewed and several trends presented.

2:35–2:50 Break

Contributed Papers

2:50

5pSA4. Methods of isolation of modal resonances (a review). Naum Veksler (Inst. of Cybernetics, Akadeemia 21, EE-0026 Tallinn, Estonia), Jean-Mark Conoir, Jean-Louis Izbicki, and Pascal Rembert (Univ. Havre, 76610 Le Havre, France)

The analytical methods of isolation and description of the resonance components of partial modes in scattering problems of acoustic, elastic, and electromagnetic waves are set forth. The results of these methods' applications are presented and their effectiveness is discussed. For some particular cases the exactness of the results obtained can be estimated as well. The following methods are considered: gradient of the phase, utilization of impedance type intermediate backgrounds, Argand diagram method, and pole evaluation on the frequency complex plane for a fixed mode order. The methods can be even used for rather complicated situations: for very narrow and rather broad resonances, near the point of intersection of the dispersion curves of phase velocities, and for very close situated resonances. The illustrative examples are given for the problems of scattering of bodies of spherical and cylindrical shape. The resonances are isolated for waves of different physical nature: diffracted (Franz type), normal (Lamb type), shear, Rayleigh type, and whispering gallery. With the properly computed modal resonances one can immediately obtain the dispersion curves of the phase and group velocities of every wave, and construct the acoustic spectrogram of the scatterer. Jointly with the well-known experimental methods, the described ones form the basis for ultrasonic spectroscopy.

3:05

5pSA5. Fuzzy structure mass per unit frequency and its asymptotic tendencies. Allan D. Pierce (Dept. of Aerospace and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

The fuzzy structure theory concept of modal mass per unit natural frequency is explored for an elastic substructure attached to a master structure. The passive influence on the master structure vibration is specified by the driving point impedance of the substructure, which in turn is expressed as a sum over terms, each corresponding to a natural mode. Term parameters are a modal mass in the numerator and the natural frequency. The fuzzy structure idealization replaces the sum by a derivative term (dissipative part) plus an integral (reactive part), both involving modal mass per unit frequency. Generic continuum models of the attachment are inhomogeneous rods, beams, and plates. (In the latter cases, the impedance becomes a matrix, with generalized forces incorporated that correspond to torques, and generalized velocities that are angular velocities.) Asymptotic ideas analogous to those that yield modal density predictions yield predictions of the mass per unit natural frequency (and its generalizations) in the limit of high frequencies. [Work supported by ONR.]

3:20

5pSA6. Continuous structures as "fuzzy" substructures. M. Strasberg (David Taylor Model Basin, NSWC, Bethesda, MD 20084-5000)

It is by now well known that the resistive part of the combined driving-point impedance of a collection of sprung masses with closely spaced frequencies of antiresonance can be expressed by the simple relation $R=1/2\pi^2 f^2 m_0$, where m_0 is the resonant mass density, defined so that $m_0 df$ is the combined mass of the sprung masses in antiresonance in a differential frequency band of width df centered at the cyclic driving frequency f. To apply this relation to a continuous or complicated structure, it is necessary to determine how the value of m_0 varies with frequency for a set of sprung masses duplicating the driving-point impedance of the structure. The procedure is illustrated by calculating m_0 for several simple continuous structures, including a long uniform rod in axial vibration, and a long uniform beam or large thin plate in flexural vibration. The values of resistance obtained by treating these systems as fuzzy structures agree with well-known values.

3:35

5pSA7. Interaction of structural and acoustical waves at a wedge-shaped junction of two fluid-loaded plates. Andrew N. Norris (Rutgers Univ., Piscataway, NJ 08855-0909)

Two semi-infinite elastic plates are joined along a line forming a wedge structure with unilateral fluid loading in the sector of angle 2Φ. The structure is modeled using thin plate theory. allowing freely propagating flexural and longitudinal waves. The junction is mechanically connected with an applied force and moment acting there to simulate a possible internal connection. The general 2-D solution is described for incidence of time harmonic structural or acoustical waves. The method of Osipov is used to express the total pressure as a Sommerfeld integral, the integrand comprising Malyuzhinets functions and particular solutions of certain difference equations. The junction conditions reduce to a system of eight linear equations. Numerical examples indicate the coupling between the modes for different wedge angles, specifically $\Phi=112.5°$, $135°$, and $157.5°$, for steel plates in water. Acoustic plane wave incidence on the flatter junction ($112.5°$) is converted almost equally, in terms of energy, among diffracted flexural and longitudinal waves. The coupling to flexural energy increases with the wedge angle, at the expense of the longitudinal energy which vanishes as $\Phi\rightarrow 180°$. An incident longitudinal wave generates relatively little acoustic sound for all values of Φ considered, with most of its energy redistributed among structural modes. The acoustical diffraction is generally greater for flexural incidence. [Work supported by ONR.]

3:50

5pSA8. Effects of component interfacial boundary conditions on component mode synthesis estimates for natural frequencies and modes. Jerry E. Farstad (Dept. of Mech. Eng., Univ. of Louisville, Louisville, KY 40292) and Rajendra Singh (Ohio State Univ., Columbus, OH 43210)

Several formulations for component mode synthesis have been proposed in the literature, with the principal differences among them being the types of boundary conditions imposed on individual components at the interfaces where they later will be joined when component natural frequencies and modes are computed. Formulations based on component modes with fixed interfacial conditions, free interfacial conditions, and mixtures of fixed and free conditions have be en proposed. While all formulations yield reliable results when all component modes are included, modal truncation errors are introduced if incomplete sets of component modes are used. In this communication, a new procedure for determining modal truncation errors in modal synthesis formulations is presented and applied to two formulations, one with mixed interface conditions and one with fixed interface conditions. Errors in estimated natural frequencies and local errors in mode shapes are investigated for the case of a two component joined beam system. The results indicate that formulations using component modes with free interface conditions are inherently susceptible to modal truncation errors because of nonzero forces and moments acting at the joints after assembly.

4:05

5pSA9. Modal impedances for nonaxisymmetric vibrations of a thin spherical shell. Richard B. Evans (Science Applications Intl. Corp., 21 Montauk Ave., Ste. 201, New London, CT 06320)

Modal impedances are found by solving the equations of motion of a thin spherical shell in terms of nonaxisymmetric modes of vibration. These mechanical impedances relate the modal expansion coefficients of the applied force per unit area and the velocity of the shell. The application and interpretation of these modal impedances require an awareness of the lack of orthogonality of certain modes. In proper combinations, the modal impedances in the nonaxisymmetric case are found to be identical to known modal impedances in the axisymmetric case. The analysis is restricted to extensional effects of a thin shell with radial forcing. An application of the modal impedances is given in the prediction of the radiation pattern of a three-dimensional array of closely spaced interacting spherical shells.

4:20

5pSA10. Investigation of the dynamic response and stability of a Stirling cycle cryocooler. F. W. Hausle, W. G. Gully (Cryogenics Dept., Ball Aerospace and Technol. Corp., Boulder, CO 80301), and Leonard J. Bond (Univ. Colorado, Boulder, CO 80309 and Denver Res. Inst., Denver, CO 80208)

Existing analytical models for acoustic cryocooler vibration analysis assume 1-D axisymmetric behavior. A multiple scales perturbation analysis is used to describe the lateral dynamic response and stability of an axially oscillating Stirling cycle cryocooler core. The model of the oscillating core mass is assumed to possess two degrees of lateral motion freedom and the forcing function is multiple harmonic axial motion. A pair of mechanical springs which support the core mass is assumed to have axial-displacement-dependent lateral stiffness characteristics which results in parametric excitation of the lateral degrees of freedom. The effect of perturbing gas–fluid forces acting on the piston, which disturb the otherwise axisymmetric motion, is investigated. The perturbation analysis predicts the dynamic response in closed form and identifies regions of operation for unstable/chaotic motion. The results of the closed-form dynamic response predictions are compared with a numerical time-step Runga–Kutta method [F. Hausle, J. Acoust. Soc. Am. **94**, 1853 (1993)]. Model results are also

compared with experimental displacement-time-frequency data obtained with a new design of cryocooler. The insight given by this analysis is expected to lead to improved cryocooler design based on better understanding of the domain of configurational and operational parameters.

4:35

5pSA11. Explicit solution to the Kirchhoff integral formulation. Sean F. Wu and Qiang Hu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

An explicit solution to the Kirchhoff integral formulation for predicting acoustic radiation from a vibrating object is derived. The radiated acoustic pressure is shown to be expressible in terms of integrations of the normal and tangential components of the particle velocity over a surface that encloses the object. If this surface coincides with that of the vibrating object, then the normal component of the particle velocity is equal to that of the surface velocity, which is normally assumed given. The tangential component of the particle velocity, however, is different from that of the surface velocity, but is determinable experimentally by using an intensity probe. For a class of special cases in which the object dilates uniformly, the tangential component of the particle velocity is identically zero. Under this condition, the radiated acoustic pressure can be obtained by directly integrating the normal component of the surface velocity over the vibrating surface, rather than solving the surface acoustic pressure first, and then the radiated acoustic pressure, as it is traditionally done in the numerical solutions to the Kirchhoff integral formulation.

FRIDAY AFTERNOON, 17 MAY 1996 REGENCY D, 1:00 TO 4:00 P.M.

Session 5pSC

Speech Communication: Perception by Special Populations

Susan L. Hura, Chair

Department Audiology and Speech Sciences, Purdue University, 1353 Heavilon Hall, West Lafayette, Indiana 47907-1353

Contributed Papers

1:00

5pSC1. The perception of stop consonants in silent-center syllables by listeners with high-frequency hearing loss. Janet W. Stack (Commun. Disord. Program, Univ. of Virginia, P.O. Box 9022, Charlottesville, VA 22906-9022)

This study investigated whether stop consonant perception in listeners with high-frequency hearing loss could be improved by eliminating nonsimultaneous masking effects of high-amplitude vowels on consonants in consonant–vowel–consonant (CVC) syllables. Full syllables (FS) contained the consonants /b,d,g/ in all combinations with eight vowels (/i,ɪ,ɛ,æ,ʌ,ɑ,u/). Syllables were produced by a male speaker. Silent-center (SC) syllables were created by attenuating vowel steady-state portions to silence. Stimuli were presented to 12 normal-hearing (NH) and 36 hearing-impaired (HI) adults under the age of 60. HI listeners had steeply sloping high-frequency hearing losses. FS and SC conditions were presented at 50 and 80 dB HL. All subjects showed more accurate vowel and consonant identification in the FS vs SC condition and at 50 vs 80 dB. Errors were significantly higher on initial than on final consonants. Acoustic analysis suggested the possibility of both temporal and spectral explanations for this final consonant advantage. In spite of experimental failure to improve consonant perception in the HI listeners, suggestions for future research emerged which might enhance the effect of the SC condition and the decrement of nonsimultaneous masking effects. [Work supported by Veterans Administration.]

1:15

5pSC2. Stimulus variability and spoken word recognition: The effects of age and hearing impairment. Mitchell S. Sommers (Dept. of Psych., Washington Univ., Campus Box 1125, St. Louis, MO 63130)

Three experiments were conducted to investigate the effects of age and hearing loss on the ability to maintain perceptual constancy in spoken word recognition. Specifically, the studies examined how variations in talker characteristics, speaking rate, and overall amplitude affected perceptual identification in normal-hearing young (NHY), normal-hearing elderly (NHE), and hearing-impaired elderly (HIE) listeners. The three dimensions were selected because variations in voice characteristics and speaking rate affect features of speech signals that are important for word recognition while overall amplitude changes do not have direct effects on phonetic identification. Thus, both phonetically-relevant and irrelevant sources of variability were investigated. Age differences, as indicated by greater effects of variability for the NHE compared with the NHY listeners, were observed for conditions with trial-to-trial variations in talker characteristics and overall amplitude. Effects of hearing impairment, as indicated by reduced scores for the HIE compared with the NHE group, were observed for conditions with variations in either speaking rate or talker characteristics. Considered together, the findings suggest that age-related changes in perceptual normalization, selective attention, and absolute sensitivity may all contribute to the reduced speech understanding often reported for older adults. Several possible mechanisms for the age- and hearing-related deficits are discussed.

1:30

5pSC3. *F0* discrimination and concurrent vowel identification by hearing-impaired listeners. Van Summers and Marjorie R. Leek (Army Audiol. and Speech Ctr., Walter Reed Army Med. Ctr., Washington, DC 20307-5001)

Labeling of concurrently presented vowels generally improves with the *F0* difference between vowels (up to a limit of 2 to 4 semitones difference). One interpretation of this finding is that the processing of *F0* differences between vowels allows listeners to group spectral components according to a common fundamental and thereby improve identification. However, the benefit of *F0* differences may be reduced for hearing-impaired listeners to the extent that accuracy in processing *F0* is compromised by the sensory impairment. In this study, normal-hearing and hearing-impaired listeners were tested to determine *F0* difference limens for synthetic tokens of five steady-state vowels. The same stimuli were then used in a concurrent-vowel labeling task. Preliminary results indicate accurate processing of *F0* differences between vowels, with thresholds generally below 2.0 Hz across vowels and listeners. On concurrent-vowel data collected thus far, hearing-impaired listeners showed an overall reduction in labeling accuracy relative to normal-hearing listeners. However, the benefit to labeling associated with *F0* differences between vowels was of similar magnitude across groups. Additional subjects are being tested and analyses correlating performance on the *F0* task with *F0*-benefit on the labeling task will be reported. [Work supported by NIH.]

1:45

5pSC4. Vowel recognition by cochlear implant patients using a large database of speakers. Philipos C. Loizou, Michael F. Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-0102), and Andreas S. Spanias (Arizona State Univ., Tempe, AZ 85287-7206)

There have been many reports on the recognition of vowels by cochlear implant patients fitted with various devices. Most of these reports, however, used a single male and/or female talker to test vowel recognition. This paper presents results on the recognition of vowels produced by a large number of men, women, and children [Hillenbrand *et al.*, J. Acoust. Soc. Am. **97**, 3099–3111 (1995)]. The aim of this study was to investigate why certain groups of speakers, e.g., children, and certain vowels are harder for the cochlear implant patients to identify. A total of 5 Ineraid patients who had been fit with continuous interleaved sampling (CIS) signal processors were used in this study. Patients' performance on vowel recognition ranged from 54% to 84%. The sex and age of the talker (that is, the talker's vocal tract size) seemed to have a significant effect on the performance for some of the patients, and in particular the "poor" patients. Individual patient's results on vowel recognition, and a detailed analysis of the vowel confusions will be presented.

2:00

5pSC5. Hearing impairment and same–different reaction time. Philip F. Seitz (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001) and Brad Rakerd (Michigan State Univ., East Lansing, MI 48824-1212)

Reaction time (RT) studies provide information about perceptual and post-perceptual information processing. A prior investigation found only small differences between the mean RTs of hearing-impaired (HI) and normal-hearing (NH) subjects performing a simple RT task with subjectively loud and soft tonal stimuli. The present study extended that group comparison to a choice RT task. Subjects with early-onset, moderate-to-severe sensorineural hearing impairments (N=8) and NH controls (N=14) made same–different judgments about digit pairs. In an auditory condition spoken digits were presented at two levels of loudness, representing the endpoints of a subject's dynamic range. In a visual condition digits were shown at the dimmest and brightest settings of a computer monitor. All subjects performed accurately in both modalities (>90% correct). Notable findings regarding RT were as follows: (1) The intensity variations had no significant effect for either group; (2) the two groups had comparable RTs in the visual condition; (3) the HI group was significantly slower in the auditory condition; and (4) subjects' auditory and visual performance were strongly correlated for the NH group but not for the HI group. Factors affecting group and individual differences will be discussed. [Work supported by NIH-NIDCD.]

2:15

5pSC6. Effects of bilingualism on non-native phonetic contrasts. Janet Calderón and Catherine T. Best (Dept. of Psych., Wesleyan Univ., Middletown, CT 06459 and Haskins Labs., 270 Crown St., New Haven, CT 06511)

Monolingual listeners often have difficulties discriminating non-native contrasts. The Perceptual Assimilation Model (PAM) [C. T. Best, G. W. McRoberts, and N. M. Sithole, JEP:HPP **14**, 345–360 (1988)] predicts poor discrimination if two non-native sounds assimilated equally well to a single native category (SC), good discrimination if a category goodness (CG) difference is perceived, and excellent discrimination if sounds are assimilated to two different categories (TC). However, implications for bilingual listeners are uncertain. Do bilinguals maintain two separate phonological systems, or do the systems interact to enhance or inhibit non-native discrimination? In this study, monolingual English and bilingual Spanish/English listeners discriminated and categorized Spanish [b–p], English [b–p^h], and three Xhosa bilabial stop contrasts. Both groups discriminated English [b–p^h] excellently, as did bilinguals for Spanish [b–p], which monolinguals discriminated less well, consistent with CG assimilation. Compatible with PAM, monolinguals showed excellent TC performance on Xhosa unaspirated/aspirated stops; bilinguals' lower discrimination indicated CG assimilation. On Xhosa implosive/pre-nasalized stops, monolinguals showed CG assimilation and greatly outperformed bilinguals, who showed an SC pattern. Conversely, on Xhosa implosive/plosive stops monolinguals showed SC assimilation; bilingual discrimination was higher, showing GC assimilation. [Work supported by NIH.]

2:30

5pSC7. Assimilation of non-native vowel contrasts to the American English vowel system. Catherine T. Best (Dept. of Psych., Wesleyan Univ., Middletown, CT 06459 and Haskins Labs., 270 Crown St., New Haven, CT 06511), Alice Faber (Haskins Labs.), and Andrea Levitt (Wellesley College, Wellesley, MA and Haskins Labs.)

The perceptual assimilation model (PAM) [Best *et al.*, JEP:HPP **14**, 345–360 (1988)] predicts that two non-native sounds that are assimilated to the same nature category will be harder for listeners to discriminate between than sounds that are assimilated to two different native categories (TC contrasts); how difficult will depend on whether they are equally good (or bad) exemplars of the single native category (SC contrasts) or not (CG contrasts). PAM was based on studies of consonant perception, as were subsequent tests of the model. This study extends the model to non-native vowels. American listeners performed keyword identification [W. Strange and T. L. Gottfried, J. Acoust. Soc. Am. **68**, 1622–1625 (1979)] and categorical AXB discrimination tasks using six non-native vowel contrasts, Norwegian /i–y/, /i–ʉ/, French /o–õ/, /œ–y/, /œ–ə/, and Thai /ə–ɯ/. Assimilation patterns for a particular vowel contrast, inferred from keyword results, were more variable than in consonant studies but nonetheless strongly related to discrimination performance: TC contrasts were better discriminated than CG contrasts, which in turn were better discriminated than SC contrasts. Moreover, listeners who assimilated a particular contrast in TC fashion were better able to discriminate it than listeners who assimilated it in CG or SC fashion. [Work supported by NIH.]

2:45

5pSC8. Perceptual assimilation of German vowels by American listeners: Context and speaker effects. Winifred Strange (Dept. of Commun. Sci. and Disord., BEH 255, Univ. of S. Florida, 4202 E. Fowler Ave., Tampa, FL 33620-8150), Ocke-Schwen Bohn (Kiel Univ., Kiel, Germany), Sonja A. Trent, Melissa C. McNair, and Katherine C. Bielec (Univ. of S. Florida, Tampa, FL 33620)

This study investigated American English (AE) speakers' perceptual assimilation of North German (NG) vowels spoken in 5 CVC contexts by four adult male speakers. Fourteen NG vowels were produced in /bVp, bVt, dVt, gVt, gVk/ syllables in the sentence "Ich habe CVC gesacht." Twelve monolingual AE listeners were tested on each speaker's corpus; consonantal context was a within-subjects variable. Response categories were indicated by /hVd/ key words and IPA symbols. Listeners responded by selecting the AE category containing the vowel most similar to the one in the utterance and rating its "goodness of fit." The percentages of selection of the modal AE response category for each NG vowel ranged from 99% for /i/→/i/ (best fit) to 41% for /ø/→/u/ (worst fit). NG front and back rounded vowels were assimilated to AE back rounded vowels. However, for seven vowels, modal category percentages differed by greater than 15% across the five consonantal contexts. There was also significant variation in assimilation patterns across the four speaker groups for 11 of the NG vowels. These results have implications for theories of L2 speech learning. [Work supported by NIDCD.]

3:00

5pSC9. Acquisition of non-native vowel categories. John Kingston, Christine Bartels, Jose Benki, Deanna Moore, Jeremy Rice, Rachel Thorburn, and Neil Macmillan (Linguist. Dept., South College, Univ. of Massachusetts, Amherst, MA 01003)

Is the acquisition of foreign phoneme identification determined by native abstract categories [Best, *Development of Speech Perception*, pp. 167–224 (1994)] or by particular tokens [Pisoni *et al.*, *Development of Speech Perception*, pp. 121–166 (1994)]? If Best is correct, members of all in-

stances of the same phonological contrast should be equally easy to identify. If Pisoni *et al.* are correct, then listeners should not generalize to new speakers or contexts. The acquisition by American English listeners of the potentially four front rounded German vowels /y:,y,oe:,oe/ was examined. Speaker and consonantal context were manipulated. Listeners improved with training. Our listeners, in contrast to those of Pisoni *et al.* (1994), generalized to new speakers and contexts. Contrary to Best's 1994 prediction that identification of the same contrast would be equally easy, it was found in this study that listeners who heard one set of speakers identified the mid tense:lax /oe:,oe/ pair more accurately than the high /y:,y/ pair, and identified the lax high:mid /y,oe/ pair more accurately than the tense /y:,oe:/ pair. Listeners who heard a second set of speakers reversed these inequalities. Thus assimilation makes generalization to novel stimuli possible but does not preclude sensitivity to phonetic differences. [Work supported by NIH Grant No. 5-R29-DC01708 and NSF Grant No. DBD92-12043.]

3:15

5pSC10. Preferred onsets. Ian Maddieson (Dept. of Linguist., UCLA, 405 Hilgard Ave., Los Angeles, CA 90096-1543)

Previous analysis of segment inventories (e.g., by Maddieson, 1984 and Lindblom and Maddieson, 1988) has shown that, across a large sample of languages, the segments that more commonly occur contrastively are those that are less complex on an intuitive scale of simplicity. But a segment type within any given language may occur in many or few words: segments that occur in more items may also be those that are in some sense more simple, or those that combine best with other segments in sequences. Within-language frequency of usage may therefore serve to show that segments of (near-) equal frequency in terms of mere membership in inventories in fact show different preference patterns. However, because there are many idiosyncratic properties of individual languages, it is important to look across a range of languages to reach general conclusions. The present paper reports on the relative frequency of the different onset consonants found in the universally present syllable type CV in a global sample of 31 languages (10 from Asia, 8 from the Americas, 6 from Africa, 3 from Europe, and 4 from the Pacific area). The counts are based on lexical frequency. [Work supported in part by NIH.]

3:30

5pSC11. Modeling the categorization of a large $F1-F2-F3$ continuum by Finnish listeners. Terrance M. Nearey and Michael Kiefte (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 0A2, Canada)

A continuum of 972 vowels was synthesized, each of which was 115 ms in duration with a falling $F0$ contour (125–100 Hz). $F1$ ranged (in 0.5-Bark steps) from 250 to 760 Hz, $F2$ from 750 to 2260 Hz and $F3$ from 1360 to 3080 Hz. $F4$ and $F5$ were fixed at 3500 and 4500 Hz, respectively. (Constraints were placed on formant separations to ensure relatively natural stimuli.) This continuum was previously categorized by a group of English speakers [T. M. Nearey and M. Kiefte, J. Acoust. Soc. Am. **96**, 3284(A) (1994)]. The same stimuli were presented to 14 speakers of Finnish at the University of Turku. Subjects categorized these stimuli with labels for the nine Finnish vowels, / i, e, y, ö, æ, u, o/. Results of logistic regression analysis will be presented and compared with previous findings for English. [Work supported by SSHRC. The assistance of Ulla Fisher and Dr. Olli Aaltonen of the University of Turku in collecting the Finnish data is gratefully acknowledged.]

3:45

5pSC12. Arabs' stress placement in English: Intelligibility test. Fares Mitleb (English Dept., Yarmouk Univ., Irbid, Jordan)

This study was intended to give an impressionistic description of Arabs' performance in English stress placement. Two native British English speakers were asked to evaluate the productions of English words by six Arabs. Findings indicated that the Arab subjects' "conscious knowledge" of a stressful speaking situation was less than satisfactory. Listeners reported a great percentage of confusion in stress placement. A number of explanations were attempted for the intelligibility problem which arises from the misplacement of stress in the subjects' productions. It was also inferred from the confusion Arabs made that suprasegmentals should be given emphasis in the teaching of foreign languages and should be taught simultaneously with segmental structure of the target tongue. [Research supported by Yarmouk Univ.]